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OF THE
ACOUSTICAL SOCIETY OF AMERICA

NOVEMBER 30 AND DECEMBER 1, 1931

Lecture Room—Physics Building
Case School of Applied Science
Cleveland, Ohio

MONDAY, NOVEMBER 30, 1931

10:00 A.M.

1. **The Application of Lamps for Sound Picture Recording and Reproducing,** R. E. FARNHAM, *Nela Park Laboratories, General Electric Company,* Cleveland, Ohio.

2. **Incandescent Electric Lamps for Sound Recording and Reproduction, and their Characteristics,** H. I. WOOD, C. SEVERIN, and E. M. WATSON, *Nela Park Laboratories, General Electric Company,* Cleveland, Ohio.

3. **Demonstration of Acoustic Properties of Rochelle Salt,** C. B. SAWYER and A. L. WILLIAMS, *Brush Laboratories Company,* Cleveland, Ohio.

4. **A Theoretical Expression Relating Loudness and Intensity,** L. B. HAM, *New York University,* and J. S. PARKINSON, *Johns-Manville Corporation.*

   Three series of experiments (see also abstract 8, Journal of Acoustical Society of America, 3, p. 7, July 1931) were performed with untrained observers as follows: (A) the sound energy was reduced by definite steps, (B) the loudness was reduced to definite fractional values of original, (C) the loudness was increased by definite multiple values of original.

The reciprocals of the remaining fractions of original loudness are plotted as ordinates y against reductions of sound energy in decibels as abscissae x, while in series C the multiple increases in loudness are plotted as ordinates y against increases of sound energy as abscissae x.

An analysis of these curves shows that they may be represented by the equation:

$$y = e^{nx}.$$

A discussion of the curves as related to this equation is given. For the present, $n = .076$ is chosen as the best value for estimating loudness reductions.

If (y) is expressed as the percentage reduction, the equation takes the form:

$$y = 100 - 100e^{-.076x}.$$

The Weber-Fechner law of loudness is a first approximation to the theoretical considerations above.

A scale for noise measurement is suggested using multiple loudness units based on the above formulae.

MONDAY, NOVEMBER 30, 1931

2:00 P.M.

1. **The American Institute of Physics,** HENRY A. BARTON, *Director of the Institute.*

   The American Institute of Physics has recently been organized through the cooperation of several scientific societies interested in the physical sciences, and the Acoustical Society of America is one of these organizing societies. By invitation of the officers of the Society, Dr. Barton, the Director of the Institute, will explain the organization, purposes, and methods of procedure.

2. **The Application of the Norris-Andree Method of Reverberation Measurement to Measurements of Sound Absorption,** R. F. NORRIS, *C. F. Burgess Laboratories,* Madison, Wisconsin, and C. A. ANDREE, *University of Wisconsin,* Madison, Wisconsin.

   Several methods of measuring sound absorption are in use but require rather large samples. The Norris-Andree method of reverberation measurement theoretically lends itself to sound absorption measurement, permitting relatively small samples to be used and necessitating no reference absorption material.

   The paper outlines in detail the mechanical application of this method to sound absorption measurement and presents data collected over a year's time in making such measurements.

3. **The Acoustic Design of Severance Hall,** DAYTON C. MILLER, *Case School of Applied Science,* Cleveland, Ohio.

   Severance Hall, the permanent home of the Cleveland Symphony Orchestra, was completed in February, 1931. From the very beginning of the plans, the donor, the Management of the Orchestra, the University and the Architects, have all been united in purpose that the acoustic condition should be as nearly perfect as is possible and that nothing should be allowed to interfere with this end. Calculations were made for proposed designs and treatments, and the architects were thus enabled to adjust and adapt plans and methods without any inconvenience, and were successful in choosing dimen-

sions, shapes and materials which in connection with the furnishings and the audience would produce the desired acoustical effect. Perhaps this adjustment has been carried out more systematically and more effectively in Severance Hall than in any other auditorium of this type.

At the conclusion of one of the sessions of the Society there will be an opportunity for the members to visit Severance Hall.

4. **The Weighting of Sound Insulation Data Against Frequency,** IRA G. ROSS, *United States Gypsum Company*, Chicago, Illinois.

Any specification of the sound insulating efficiency of a partition or other building unit must express, in terms of normally generated sounds and human hearing, its insulating value over the whole range of normal sound. Attention is called to the fact that, in making such an evaluation of the insulating efficiency, the Reduction Factors over the normal frequency range should be weighted to take account of the manner of distribution through the frequency spectrum of the peak loudnesses of normal sounds, the acuity of hearing and the normally maintained level of incidental noise. Curves of uniform insulation efficiency in terms of Reduction Factor at each frequency are derived from the data on peak loudnesses, acuity of hearing and incidental noise.

It is frequently desirable to reduce the overall insulation value of a partition to expression in a single figure. This has been done in the use of "Average Reduction Factor." It is suggested, in view of the considerations above, that a more representative expression might be made in terms of minimum insulation value referred to the uniform insulation levels as derived.

5. **The Determination of Absorption Coefficients for Frequencies up to 8000 Cycles,** F. L. HOPPER, *Electrical Research Products, Inc.*, Los Angeles, California.

The necessity for the determination of coefficients of sound absorption for frequencies up to 8000 cycles has been brought about by the improvements being effected in increasing the range of recorded and reproduced frequencies in sound pictures and broadcasting. Naturalness and intelligibility are enhanced by the addition of these higher frequencies which further increases the effect of realism. Consequently, with the development of apparatus capable of handling a wider frequency band, a knowledge of acoustical conditions over an equally wide band of frequencies is required.

From a knowledge of absorption coefficients at the lower frequencies it appears to be impracticable to predict coefficients at the higher frequencies, consequently a resort to measurements is the only satisfactory solution. The apparatus and technique for the determination of coefficients of absorption for frequencies up to 8000 cycles by the chamber reverberation method are discussed. Factors affecting these measurements, such as sound absorption in air, size and methods of mounting the test samples, and so on, are described. The absorption coefficients for a number of commercial materials for these higher frequencies are shown. Since materials having different types of frequency absorption characteristics, as well as absolute absorption, are required for such work as: the reduction of noise, the acoustical treatment of theatres and auditoriums, and the treatment of rooms or stages used for the electrical recording of speech or music, such information is necessary in order to intelligently design such acoustical treatment.

6. **Reverberation Characteristics of Sound Pictures Sets and Stages,** R. L. HANSON, *Bell Telephone Laboratories*, New York, N.Y.

In this paper are presented the results of reverberation measurements made in about sixty motion picture sets and sound stages together with a theoretical consideration of the behavior of residual sound in motion picture sets. The paper points out that, on the basis of the experimental results and the theoretical considerations a uniform decay of the intensity level of residual sound in a motion picture set having large areas opening into a sound stage would not be expected. The initial decay rate is approximately that which would exist in the set located in the open, but changes with time and approaches that which would normally exist in the stage itself. The detrimental effect which such characteristics have on the quality of the reproduced sound depends upon various factors such as the volume range which can be accommodated by the system, the relative intensities to be recorded, the type of set and stage, etc.

7. The Acoustics of Large Auditoriums, S. K. WOLF.

Laws of acoustics known to apply to smaller volumes may not always be taken as true for large auditoriums. For example, computed reverberation times are radically different from actual measured times. In the case of Madison Square Garden, 6,200,000 cubic feet in volume, the calculated time of 35 seconds at 500 cycles contrasts with a measured time of 7.65 seconds. Use of the formula recently proposed by Knudsen reduces the discrepancy, but insufficiently. At 2000 cycles the computed time was 13 seconds; whereas, the measured value was 5.4 seconds. Some general possible causes of the anomaly are suggested.

TUESDAY, DECEMBER 1st, 1931

10:00 A.M.

1. The Effect of Form on the Reverberation of Sound in Rooms, VERN O. KNUDSEN, *University of California at Los Angeles*, Los Angeles, California.

Measurements of the *mean free path* p of sound in three dimensional models of eleven typical auditoriums—such as rectangular rooms (with and without balconies), fan shaped theatres, country and city churches, and rooms with gabled, barreled or domed ceilings—indicate that p varies from about 3.8 V/S (V = volume, S = interior surface) in a large rectangular room with a low ceiling to 4.3 V/S in a cruciform type of church auditorium. The reverberation formula for the *impressed* vibrations takes the form $t=\dfrac{kV}{-rS\log_e(1-\alpha)+4mV}$ where k depends upon the shape of the room and has a value of .046 in the large room with the low ceiling, .053 in the cruciform type of church auditorium, and about .049 for the conventional type of room; r is a probability factor which depends upon the location of the absorptive material in a room; and m is the attenuation constant of the air. The more exact theory of reverberation must describe also the rate of decay of the *free* vibrations in a room. Some measurements of reverberation have been made in a small room which exhibit gross violations of the generally accepted reverberation formula. These measurements reveal the necessity of considering the free vibrations in reverberation theory.

2. Approximations in Determining the Total Absorption of a Room, C. A. ANDREE, *University of Wisconsin*, Madison, Wisconsin.

In the formulae for the decay of sound in a room the total absorption is customarily taken equal to $\Sigma_i \alpha_i S_i$, where α_i is the absorption coefficient for any uniform area S_i. It is shown that this method of determining the total absorption is an approximation requiring the assumption "that sound which is reflected from material having a given reflection coefficient has a probability of encountering, on its *next* reflection, material with a like coefficient which probability is equal to the ratio of the area of that material to the total area present." This assumption requires in many instances an amount of "mixing" of the sound in excess of what may normally be expected from dispersion phenomena. The author suggests a method of attack which permits one to account for varying degrees of dispersion and applies this method of attack to the simple case of a cubical room. Numerical values of, $\bar{a}$, the average absorption coefficient are determined which shows the effect of assuming different degrees of dispersion, and the effect of altering the distribution of the absorbing material in the room.

The theory indicates that an error is introduced in determining the sound absorption coefficient of materials due to the uneven distribution of the materials in the average sound chamber.

3. **The Apparent Reduction of Intensity Level of Noise,** EMORY W. TAYLOR and DONALD A. LAIRD, *Colgate University*, Hamilton, New York.

Practical observations indicating that a small reduction in noise appears vastly larger than it actually is have been verified and the degree of apparent reduction measured. Naïve observers were given a 3-A audiometer stimulus to listen to, immediately following which they were to indicate the one of several other loudnesses which reduced the original stimulus by one-fourth, one-half, or three-fourths. The audiometer sound was heard in one ear, and in other phases of the study was amplified to bathe the subject, with results that are practically identical. An intensity level of 80 decibels appears to be reduced by one-half when it is only reduced by 23 decibels, etc. Lantern slide charts of results (5 minutes).

4. **The Influence of Noise Stimuli upon the Normal Secretion of Saliva and Gastric Juices,** HENRY T. GRAHAM, JAMES W. STEPHEN and DONALD A. LAIRD, *Colgate University*, Hamilton, New York.

Partial confirmation of the hypothesis that the ill effects of appropriate noises are due to the generation of a biological fear reaction was sought in the effects of noises upon these secretions. The complicated noise of the 3-A audiometer at an intensity level of 50 decibels or more was found to cut down the secretion of saliva by 50% or more. Similar noises of 40 decibels or more cut into the secretion of gastric juices. Possible methods of overcoming these effects are discussed. Lantern slides of methods and findings (8 minutes).

TUESDAY, DECEMBER 1st, 1931

2:00 P.M.

1. **A Uni-Directional Ribbon Microphone,** HARRY F. OLSON, *RCA Photophone, Inc.*, Camden, New Jersey.

Directivity has been found to be desirable in sound pickup systems to improve the ratio of direct to generally reflected sounds and to otherwise discriminate against un-

desirable sounds. The ribbon microphone is a simple directional system in which the directional characteristics are independent of the frequency. The response is a function of the angle between the normal to the ribbon and the direction of propagation of the incident sound. The ribbon microphone is a pressure gradient microphone and the response is a measure of the velocity component in a sound wave. Using a mass controlled element the velocity of the ribbon is in phase with the velocity in the sound wave. By a suitable combination of this microphone with a pressure operated ribbon microphone a uni-directional microphone may be obtained. The pressure ribbon microphone should be resistive controlled and the response is then a measure of the pressure component in the sound wave. The velocity of the ribbon in this microphone is then in phase with the pressure. By combining the outputs of two such microphones the response as a function of the angle θ between the normal to the ribbon and the direction of propagation of the incident sound is expressed by $R=R_0(1+\cos\theta)$ where R_0 is the sensitivity of the pressure and velocity ribbon microphone for $\theta=0$. Preliminary experimental results verify the theory with respect to the phase relations and the directional characteristics.

2. The Noise Survey of the Rapid Transit Lines of New York City, G. T. STANTON and J. E. TWEEDDALE, *Electrical Research Products, Inc.*, New York, N. Y.

The extension and increased operation of rapid transit facilities in the larger cities to keep pace with their growth creates a noise problem of the first magnitude. The reduction of existing noise conditions due to elevated and subway lines and the planning of future transportation facilities having a minimum of disturbance can only be effected after a complete and thorough investigation of all contributory noise sources and the various controlling factors.

In view of these considerations, this survey was initiated to study the noise conditions created by the operation of the rapid transit lines of New York City to provide information to guide in the planning of future extensions into outlying undeveloped areas, as well as to determine the effectiveness of proposed noise reduction measures on existing lines. The extensive nature of this survey involved a comprehensive study of all types of elevated and subway constructions—of the various types of cars and auxiliary equipment, roadbed, and the effect of the immediate surroundings, such as buildings and the contour of the land. Comparative data on the noise created for any given combination of conditions was obtained so that representative average results can be predicted showing the effect of the variable factors influencing the creation of noise. From the careful analysis of these results, very important conclusions, which in many cases were contrary to those generally held, were drawn which are of infinite aid in studying the economic feasibility of a constructive program of noise reduction.

General conclusions regarding elevated construction indicated that improvement of the elevated structure and roadbed as well as improved car construction and maintenance will effect a material reduction in noise. In addition, the combination of rapid transit and vehicular facilities in a comprehensive system of regional planning will aid materially in reducing the economic loss caused by these noise sources.

3. On Sound Diffraction Caused by Rigid Circular Plate, Square Plate and Semi-Infinite Screen, L. J. SIVIAN and H. T. O'NEIL, *Bell Telephone Laboratories*, New York, N. Y.

Of the three problems named above, a theoretical solution due to Sommerfeld is known for the case of the semi-infinite screen when the wave front is plane and parallel to the

edge of the screen. For the other two problems no theoretical solutions have been given. Both these cases are of interest in connection with the types of diffraction caused by a number of practical acoustic devices. Hence it was thought worthwhile to present some experimental data and to indicate a calculating procedure which, though not at all rigorous, accounts for a number of the observed facts. In the semi-infinite screen problem the theoretical solution is restated in forms suitable for numerical evaluation. Several graphs showing the sound pressure amplitudes in regions of particular interest are given.

4. On the Loudness of Noise, H. B. MARVIN, *General Electric Company*, Schenectady, New York.

In order to determine the usefulness of pure tone data in design of noise meters intended for measurement of complex sound, a reference tone of 1000 cycles at various levels was compared with noises typical of electrical machinery. The comparison was made on the basis of equal loudness determined by ear and also by noise meter. A description of the method and preliminary results are given. The results show good agreement between ear and meter in the noise levels covered by two frequency weighting networks designed from pure tone data. Need for a third network in the higher noise levels is shown.

5. Control of Pitch in Wind Instruments, JOHN BELLAMY TAYLOR, *General Electric Company*, Schenectady, New York.

A statement often heard or read as coming from some prominent musician is to the effect that of all the orchestral instruments only the violins (including the larger members of violin family) and the slide trombones can play in perfect intonation. The absence of frets on the finger-board permit performer to nicely determine vibrating length of string and sliding tubes of trombone give similar opportunity for precise determination of air column length. Coupled with the claimed pitch superiority of violins and trombones there is also an implication that a better "ear" is required for acceptable playing of these instruments than is needed for playing other orchestral instruments, viz., flutes, oboes, clarinets, bassoons, horns, and trumpets.

While the wind instruments do have a limited tuning range, it is misleading to speak of them as of "fixed pitch," and erroneous to assume that uncovering the proper vents, or depressing the proper pistons assures a tone of definite pitch. This is because the air column of instrument is, through mouthpiece or reed, sufficiently *coupled* to lips and oral cavity of player to give him an appreciable control of the air column time of vibration. This allows sounding the tone either definitely "out of tune" or, within reasonable limits, raised or lowered in pitch to come in close accord with the tones from other instruments. To exercise this pitch control musically and artistically the wind instrument player is as dependent on a good "ear" as is the performer on the strings.

6. Direct Meaurement of Sound Energy Density and Sound Energy Flux in a Complex Sound Field, IRVING WOLFF and FRANa MASSA, *RCA Victor Company, Inc.*

Measurements were undertaken to determine whether advantageous results in the elimination of the effect of interference patterns in a room could be obtained by measuring either the energy density or energy flux in a sound field rather than the pressure or velocity. The measurements of energy density were made by determining, at the same

point in space, the pressure and three velocity components at right angles to each other. By this means a calculation of the potential and kinetic energies and, therefore, the total energy density was made possible. In the course of these measurements an investigation was made of the distribution of pressure and velocity as well as energy at a series of points in the room and data are given to show the relative distributions of these quantities. A method is described for measuring the energy flux density. As a result of these measurements and others which have been made using a ventilated ribbon microphone, the conclusion was reached that the best conditions for minimizing the effect of interference pattern when measuring the output of a sound source in a specified direction are obtained by the use of a damped room and a directional microphone properly oriented so as to eliminate the most serious reflections.

7. Liveness of Rooms, R. L. HANSON, *Bell Telephone Laboratories, Inc.*, New York, N.Y.

Experience has shown that the sense of liveness of reproduced sound cannot always be correlated with the reverberation period of the region in which the sound originates. A series of experiments have therefore been carried on which show that the liveness of reproduced sound is not a function of the reverberation period but is dependent upon the irregular fluctuation of the sound intensity at the microphone. This fluctuation which often appears during sustained as well as growing and decaying tones is a function of the size, shape, and abxorbing power of the room in which the microphone and sound source are located. It explains the effect of monaural pick-up, microphone distance, etc., on liveness. The conditions which give rise to a high degree of intensity fluctuation and hence liveness are often such that a long reverberation period results. It is pointed out in the paper that if there is an optimum reverberation period for listening in a room, that period should also be optimum for recording. The intensity fluctuation due to shifting interference patterns should, in the case of a monaural pick-up pe reduced to a degree comparable to that which is apparent to a two ear listener.

TECHNIQUE OF MICROPHONE CALIBRATION*

By STUART BALLANTINE
Boonton Research Corporation

The usual purpose of microphone calibration is to determine the relation between the open-circuit voltage generated by the microphone and the pressure at the surface of its diaphragm or the pressure in a sound field. For technical purposes and with microphones having inaccessible diaphragms it is usually sufficient to compare the unknown microphone with a standard microphone; in other cases an absolute calibration may be desired. A variety of methods of absolute calibration are available but no attempt shall be made here to consider all of them. A detailed discussion of the technique of several methods which have been found convenient in practice may be presumed to be of more value to practicing engineers in view of the scarcity of publications dealing with the manipulative side of the subject.

Particular reference shall be made to the condenser microphone of the stretched diaphragm type.

1. Calibration in Terms of Diaphragm Pressure and Pressure in a Progressive Sound Wave.—Until recently, microphone calibrations have been exclusively performed by applying known alternating pressures to the diaphragm. A suitable technique, employing the thermophone and pistonphone, was worked out by Arnold, Crandall, and Wente and this early work made possible and gave impetus to a number of important quantitative investigations in acoustics and hearing which have since been made.

For the majority of uses to which the microphone is put, however, this type of calibration is of questionable significance because the pressure at the diaphragm may not be the same as the undisturbed pressure in the sound field under observation. This was discovered experimentally a few years ago as the result of an attempt to check the thermophone calibration by means of the Rayleigh disk and the discrepancy was attributed in part to the effect of diffraction around the microphone whereby the pressure on the diaphragm was increased over that existing in the undisturbed sound wave.[1] Employing a spherical microphone

* Presented at the meeting of the Acoustical Society of America, Camden, N. J., May 5, 1931; presented as part of a paper "Technique of Loud-Speaker Sound Measurements," Philadelphia Section, Institute of Radio Engineers, May 13, 1931, and at the Annual Convention, Institute of Radio Engineers, Chicago, June 5, 1931.

[1] Stuart Ballantine: "Effect of Diffraction Around the Microphone in Sound Measurements"; *Phys. Rev.*, *32*, 988, 1928; *Proc. Inst. Radio Engs.*, *16*, 1639, 1928.

mounting for which this effect could be computed, more careful comparisons with the Rayleigh disk revealed a residual discrepancy which is ascribed to acoustical resonance in the recess in front of the diaphragm.[2] The possibility of the diffractive effect was recognized by Crandall, who attempted unsuccessfully to evaluate it; both effects were suspected by Aldridge[3] and Barnes,[4] and the cavity resonance was independently discovered by West,[5] Oliver[6] and Hartmann.[7]

These effects are of considerable importance; for example, in the measurement of the pressure in a plane progressive sound wave their combined action leads to errors as high as 300 per cent at frequencies of the order of 3,000 cycles. The extent to which important older investigations in acoustics and hearing have been prejudiced by these effects in condenser microphones calibrated by means of the thermophone can best be judged by those who have been responsible for them and who are familiar with the details of the technique employed. At least some review of those results would seem desirable for the purpose of applying the proper corrections where they are found to be necessary.

An instructive physical picture of the diffractive effect may be obtained by considering the simple case of a microphone of spherical form for which the pressure increase can be calculated theoretically. The pressure ratio varies from unity at low frequencies to 2 at high frequencies. The calculated variation for spheres of various diameters are reproduced in Fig. 12.

Resonance in the cavity in front of the diaphragm produces an additional increase of pressure which also varies with the frequency. In a microphone having a $1\frac{1}{2}'' \times \frac{1}{2}''$ cavity the pressure increase amounts to about $2\times$ at 3,000 cycles.

In view of these effects it seems desirable to supplement the usual calibration of the microphone in terms of pressures applied to the diaphragm by such additional tests as will establish the relation of the response to the actual pressure in an undisturbed wave field. An extended calibration program might be drawn up along the following lines:

[2] Stuart Ballantine: "Effect of Cavity Resonance on the Frequency Response Characteristic of the Condenser Microphone" *Contributions from Radio Frequency Laboratories, No. 18,* April 15, 1930; *Proc. I.R.E., 18,* 1206, 1930.

[3] A. J. Aldridge: *Jour. Post Office Elec. Engs., 21,* 223, 1928.

[4] E. J. Barnes: *Proc. Wireless Section, I.E.E., 3,* No. 7, 59, 1928.

[5] W. West: "Pressure on the Diaphragm of a Condenser Transmitter": *Proc. Inst. Elec. Engs.,* **5,** 145, June 1930.

[6] D. A. Oliver: "An Improved Condenser Microphone for Sound Pressure Measurements"; *Jour. Scientific Inst.,* **7,** 113, 1930.

[7] C. A. Hartmann: *Elek. Nach. Tech.,* **7,** 104, 1930.

(1). Response as a function of frequency per unit pressure applied to the diaphragm.

(2). Response as a function of frequency per unit pressure in a plane progressive sound wave at normal incidence to the face of the diaphragm.

(3). From the above, the pressure ratio as a function of frequency.

(4). The directivity characteristic at different frequencies for a plane progressive wave.

Information of this type will generally be sufficient for most microphone applications.

2. Terminology.—The following terms shall be employed, some of which have been suggested in earlier papers.

Response.—This is the open circuit voltage generated by the microphone in response to unit acoustical pressure. Root-mean-square values are usually employed and a subscript may be employed to designate the polarizing voltage in the case of a condenser microphone. Thus R_{200} designates the response with a polarizing battery of 200 volts. The pressure is usually expressed in *bars* (1 bar = 1 dyne/cm^2).

Pressure Response.—The response in terms of unit sinusoidal pressure at the diaphragm.

Wave Response.—The response in terms of unit pressure in a sound wave. Unless otherwise stated the wave will be considered to be of the plane progressive type $p = p_0 \sin \omega(t - x/c)$.

Pressure Ratio.—The ratio of the pressure at the diaphragm to the pressure in an undisturbed wave field. This may be determined by taking the ratio of the wave response to the pressure response.

Directivity.—The relation between the response and the angle between the wave-normal and the normal to the diaphragm, the wave being usually of the plane progressive type. In the general case there are two directivity characteristics, corresponding to the azimuth and altitude. In the case of a spherical or symmetrical microphone only the angle of colatitude is involved.

I. Pressure Response

3. Method Employing Electrostatic Excitation.—A simple and convenient means for applying a known mechanical force to the microphone membrane is illustrated schematically in Fig. 1. This method has been in successful use for about seven years and was devised to

avoid the experimental inconvenience of the thermophone. The pressure upon the diaphragm is produced by the electrostatic force between the diaphragm and a grille-shaped electrode mounted in front of it at a small distance (0.015 to 0.040 inch). Due to the perforations in the grille and the separation employed the acoustical reaction on the diaphragm is negligible. One of the chief conveniences of this method resides in the fact that the force for a fixed voltage amplitude is independent of the frequency. The force exerted by the grille can be accurately computed from its physical dimensions and separation, so that the absolute precision of the method depends only upon the measurement of the voltages and the physical dimensions of the grille, particularly of the spacing between grille and diaphragm. In contrast with this we have, in the case of the thermophone, some fifteen or more

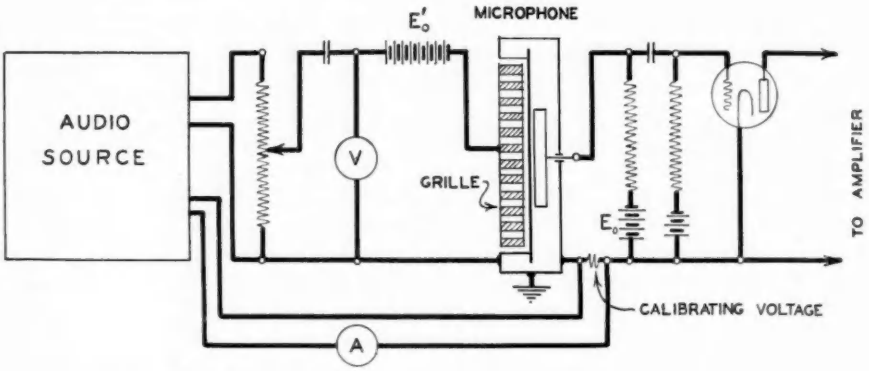


FIG. 1. *Method of pressure calibration employing electrostatic excitation.*

physical and electrical quantities to deal with, some of which (e.g. thermal conductivity of the gas and thermal capacity of the foil) are not known with satisfactory accuracy. The considerable variation of generated pressure with frequency, and the special apparatus required for the supply of hydrogen, are further disadvantages of the thermophone which render the electric grille comparatively attractive from the viewpoint of manipulative convenience.

Electrical connections for this method of calibration are shown in Fig. 1. The force exerted upon the diaphragm is proportional to the square of the voltage between grille and diaphragm and two methods of excitation may be employed.

In the usual method a d-c polarizing voltage E_0' is superposed upon the alternating voltage $E_1 \sin \omega t$, so that the force exerted is:

$$F = p = \frac{K}{8\pi d^2}\left[E_0'^2 + \frac{E_1^2}{2} + 2E_0'E_1 \sin \omega t - \frac{E_1^2}{2}\cos 2\omega t\right]. \quad (1)$$

Here d represents the distance between grille and diaphragm and K is a dimensionless factor, dependent upon the geometry, called the *geometrical constant* of the driver. E_0' should be chosen large enough relative to E_1 so that the percentage of 2nd harmonic ($E_1/4\,E_0 \times 100\%$) is negligible. Then the desired alternating pressure is:

$$p_1 = 2.64 \times 10^{-6} E_0' E_1 K/d^2 \qquad \text{(bars, volts, cms)}. \quad (2)$$

The values of p_1 will be rms values, of course, if rms values of E_1 are employed.

In the second method of excitation, the polarizing voltage is omitted and the pressure is of double frequency:

$$p_1 = 8.6 \times 10^{-7} E_1^2 K/d^2 \qquad \text{(bars, volts, cms)}. \quad (3)$$

In this case p_1 is expressed in bars (rms) and E_1 in volts (rms).

The first method of excitation will be presumed for purposes of discussion.

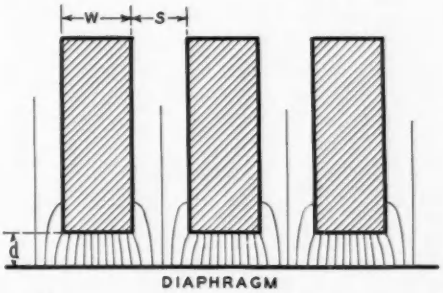


FIG. 2. *Cross section of electrostatic grille having slots deep in comparison with their width.*

4. Geometrical Constant K for a Grille Having Slots Deep in Comparison with their Width.—The geometrical factor K in (2) and (3) represents the effect of the slots in the grille in reducing the force below that which would be obtained with a plane imperforate electrode. This factor has been calculated for a grille having slots which are deep in comparison with their width and in which the bars are not too narrow in comparison with the separation of grille and diaphragm (Fig. 2). The mathematical details of this calculation will be found in the annexed **Note 1.** The calculation has been performed for an infinite structure and the fringing effects at the periphery of the diaphragm are neglected. This omission is not serious in the planar grille used with the flush type diaphragm, and may be alleviated somewhat in the recessed diaphragm

type by using a minimum of insulation (one sheet of mica) between the grille and the inside wall of the recess. Fortunately the loss of pressure at the periphery of the diaphragm is not so serious as it would be at the

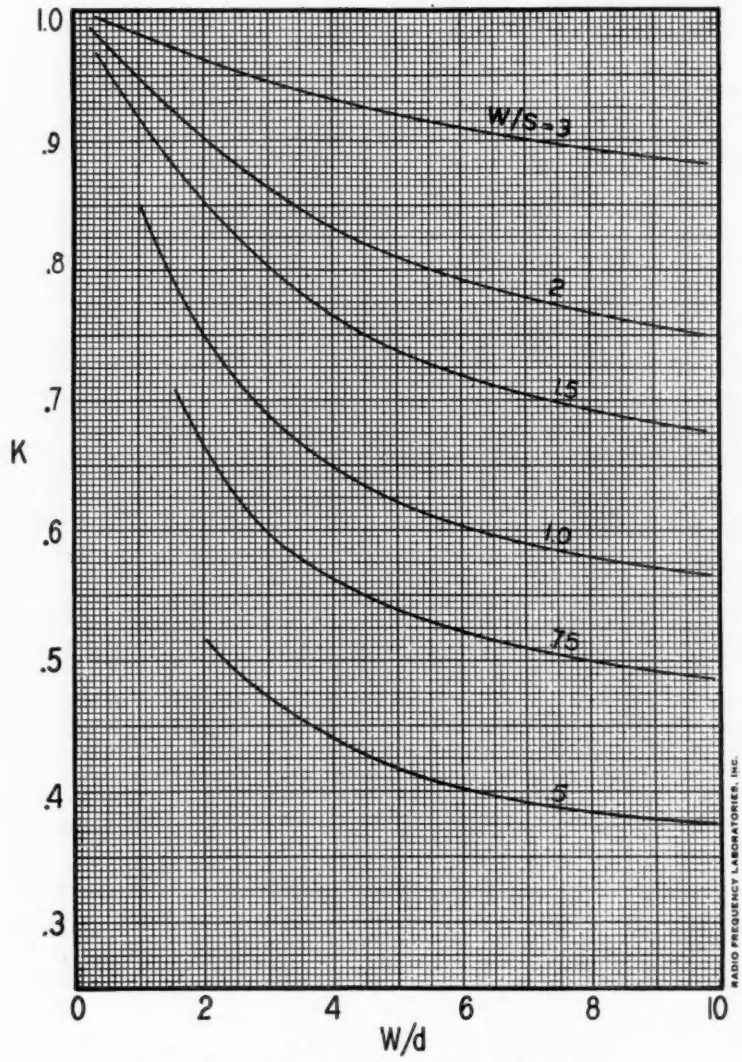


FIG. 3. *Geometric constant K for grille shown in Fig. 2.*

center. If a correction is desired the equations given by Jeans[8] for the bend in a Leyden jar may be found useful.

It is convenient to have K as a function of the dimensionless quantities W/S and W/d, where W represents the width of the bars, S the

[8] J. H. Jeans: "*Electricity & Magnetism*"; 5th Ed., p. 277 Cambridge 1925.

width of the slots and d the distance between the bars and the diaphragm. K and W are both given as functions of a single parameter ψ in equations (13) and (15), Note 1, and K may be put in the above form by cross-plotting.

The calculated values of K are shown in Fig. 3 over the ranges of W/S and W/d for which the approximations employed in the derivation of the formulae in Note 1 are valid. The dimensions of the grille may be chosen so as to be well within these ranges without embarrassment from acoustic reaction on the diaphragm. Typical dimensions which have been employed are $W=0.062''$; $S=0.031''$; $d=0.030''$ approximately.

Several types of grilles are shown in Fig. 4. Grille A, having approximately the dimensions given above, is of the planar type and suitable

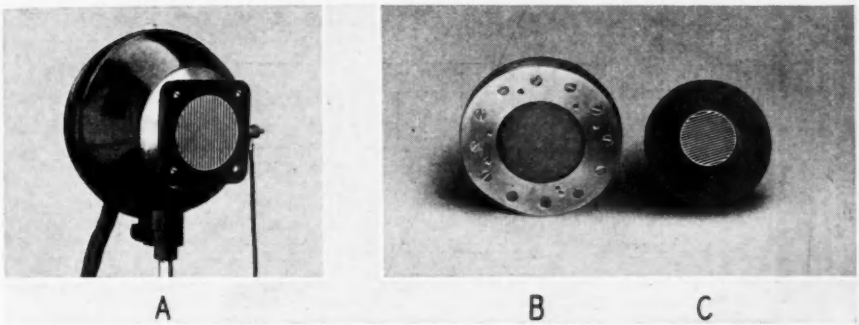


FIG. 4. *Illustrating several types of grilles.*

for use with a microphone having a flush type diaphragm.[2] Grille B is also designed for use with a flush type diaphragm and consists of a stretched wire gauze. Grille C is intended for use with a microphone of the conventional type having recessed diaphragm.

5. Effect of Insufficient Grille Area.—The grille should cover the diaphragm surface as completely as possible. Incomplete coverage has been found not only to affect the absolute value of the force exerted, but also to produce a spurious variation with frequency. This latter affect is probably caused by variations in the impedance of various annular elements of the diaphragm with frequency. At lower frequencies for example, the impedances of annular elements near the edge are considerably larger than those near the center. As the frequency increases the cushioning effect of the air-film between back-plate and diaphragm

[9] Previous citation (2), p. 8.

comes into play and the impedance tends to become more uniform. Forces applied to the outer annular elements are now comparatively more effective than at low frequencies. Experimental curves showing the effect of a 20% reduction in grille diameter are reproduced in Fig. 5. It will be observed that not only is there a shift in level but also a relative frequency variation. The response with the smaller grille is reduced 18% at 100 cycles, 31% at 1,000 cycles and 48% at 6,000 cycles. Complete coverage is easily arranged, of course, in planar grilles of the types shown at A and B Fig. 4.

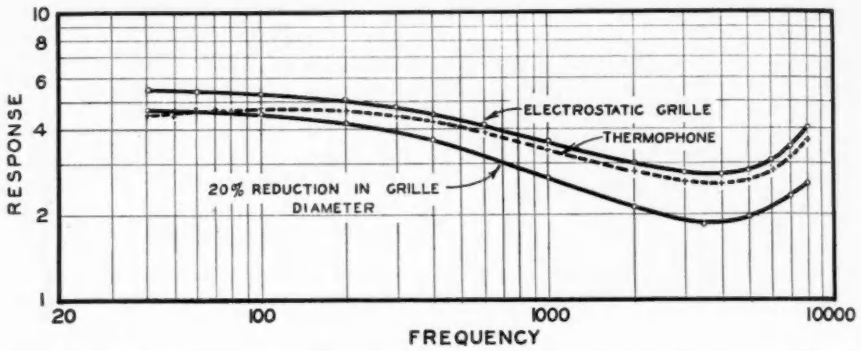


FIG. 5. *Pressure calibration of condenser microphone illustrating effect of reduction of grille area and a comparison of electrostatic and thermophone calibrations.*

6. Comparison of the Electrostatic Grille and Thermophone Methods. The results of an experimental comparison of the above method and the thermophone (gold-foil in hydrogen) are shown in Fig. 5. The agreement in frequency variation is good above 200 cycles; below this there is a departure of 6 or 7%. The absolute values differ by about 5%, which is well within expectations. In this experiment the polarizing voltage on the grille was 700 and the alternating voltage was 50 (rms). Calculation shows that the steady displacement of the diaphragm due to the electric force is negligible at $d = .020''$.

7. Thermophone Method.—In the form conventionally employed for calibration purposes the thermophone comprises one or more strips of metal foil mounted in a small enclosure of which one wall is formed by the microphone diaphragm. In the usual method the strip carries a steady current upon which a sinusoidal alternating current is superposed. The resulting variation of temperature in the enclosure generates an alternating pressure at the diaphragm which is of the same frequency as the alternating current provided the amplitude of the latter is small compared with the steady component.

The structure of a typical thermophone is illustrated in Fig. 6. The strips are ordinarily of gold foil, although platinum foil may also be employed. In the thermophone shown in Fig. 6 the mounting of the

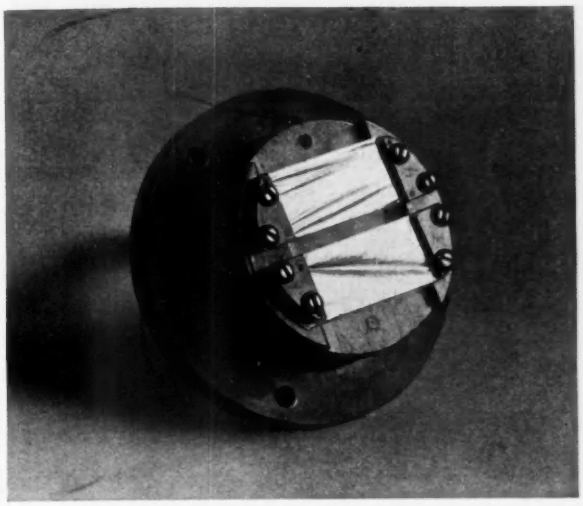


FIG. 6. *Illustrating construction of thermophone with gold-foil strips.*

gold foil strips is greatly facilitated by providing the clamping pieces with pins which are engaged by holes in the supporting blocks, thus

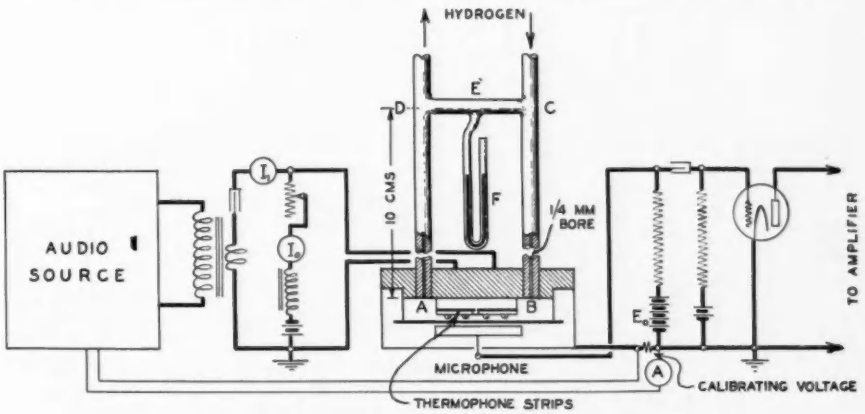


FIG. 7. *Method of pressure calibration employing the thermophone.*

avoiding tearing of the foil by movement of the clamping pieces when the screws are tightened. These pins are made from the points of ordinary sewing needles.

The thermophone enclosure is generally filled with dry hydrogen. This very slowly circulated through the enclosure by means of two capillary tubes, A and B, Fig. 7. In order that the steady pressure in the enclosure shall not differ appreciably from the standard atmospheric pressure, the circulation of the hydrogen is effected by a pressure head and exhaust head which are equal. The pressure in the enclosure is checked by means of the manometer F. The pneumatic resistances of the tubes DE and EC are equal.

The usual electric connections are also shown in Fig. 7. Ordinarily the desired calibration represents the open-circuit, or generated, voltage of the microphone for a given alternating pressure at the diaphragm. In order to obtain this from the readings of the instrument which terminates the CM amplifier, the amplifier-voltmeter is calibrated by means of a calibrating voltage e introduced in series with the microphone. This calibrating voltage is on the same footing as the generated voltage of the microphone. The transduction of the microphone is generally expressed in milli-volts per bar.

8. Theory of the Thermophone.—The thermophone was first made available as a quantitative instrument by the mathematical calculations of the generated pressure which were performed by Arnold and Crandall.[10] This theory gave a rather unsatisfactory account of the pressure at low frequencies and was further improved by Wente,[11] who considered the problem in more detail and added a term to the differential equation of heat transfer in the gas to take account of the rate of heat storage due to the pressure variation. Wente tested his theory experimentally by comparing thermophones of different physical characteristics (gold and platinum foil in air and hydrogen) and by comparison with the pressures generated by the pistonphone at low frequencies.

The formula for the generated pressure which I have been using was derived independently (1921) from substantially the same premises. The analysis differs slightly from Wente's however, in including a boundary condition to take into consideration the fact that the walls of the enclosure are generally of metal and therefore at ambient temperature. This formula is given below; a brief outline of its derivation will be found in the appended Note 2.

In the derivation of this formula it is assumed that the thermophone strips constitute a plane laminum parallel to the plane of the diaphragm

[10] H. D. Arnold and I. B. Crandall: *Phys. Rev.*, **10**, 22, 1917.
[11] E. C. Wente: *Phys. Rev.*, **19**, 333, 1922.

and at approximately equal distances from it and from the plane inner face of the mounting plug.

Under the assumptions discussed below and in Note 2, the following expression for the amplitude (peak value) of the alternating pressure developed in the thermophone enclosure is obtained:

$$p_1 = \frac{2S}{\omega m C V A \alpha} \cdot \frac{.48 I_0 I_1 R}{D^{1/2}} \tag{4}$$

where

$$D = \left(1 - \frac{4kS^2}{\omega C V A}\right)^2 + \left(1 + \frac{4S}{VA\alpha} + \frac{4kS\alpha}{\omega C} + \frac{4kS^2}{\omega C V A}\right)^2$$

and

$$A = \frac{T_a}{T_{s0}} \cdot \frac{\gamma}{\gamma - 1} - 1$$

$$m = (\gamma - 1) T_{s0} / \gamma P_0$$

$$\alpha = (\pi f C_p \rho / k)^{1/2}$$

C = total thermal capacity of the foils (mass × specific heat)
I_0 = steady component of foil current (amps.)
I_1 = peak value of a-c in foil (amps.)
R = total resistance of thermophone foils (ohms)
T_{s0} = mean temperature of the foil (absolute degrees)
T_a = average temperature of the gas (absolute degrees)
k = thermal conductivity of the gas
ρ = density of the gas
C_v, C_p = specific heats of the gas
$\gamma = C_p / C_v$
P_0 = average pressure in the enclosure (ambient)
S = total area of one side of thermophone foils
V = volume of enclosure
f = frequency of I_1
$\omega = 2\pi f$.

The values of the quantities entering into the expression for α are to be taken at T_{s0}, the average strip temperature.

The formula derived by Wente is well known but may be reproduced here for convenience of reference. Neglecting radiation and employing the above notation:

$$p_1 = \frac{2S}{m\omega CVA\alpha} \cdot \frac{0.48 I_0 I_1 RM}{\left[1 + \left(1 + \frac{2S}{VA\alpha} + \frac{4kS\alpha}{\omega C}\right)^2\right]^{1/2}}, \tag{5}$$

where

$$M = \left(1 - \frac{S_w}{V\alpha} + \frac{S_w^2}{2V^2\alpha^2}\right)^{1/2} \tag{6}$$

is a corrective factor, derived separately, to take account of the conductivity of the walls. S_w is the total area of the walls of the enclosure.

For a numerical comparison of (4) and (5) consider a specimen thermophone using hydrogen, in which the effect of wall conductivity is greater than in air. Let $k = 4.2 \times 10^{-5}$, $\alpha = 1.45 f^{1/2}$, $S = 7$ cm^2, $V = 8.5$

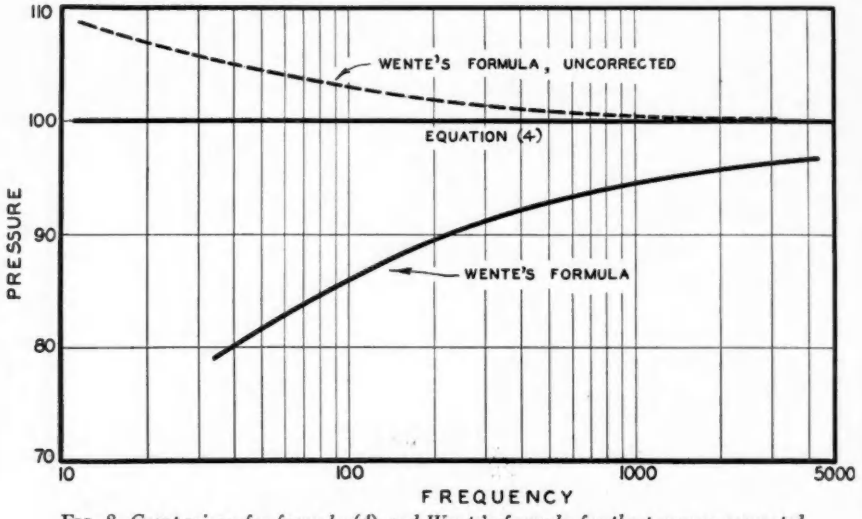


FIG. 8. *Comparison for formula (4) and Wente's formula for the pressure generated by the thermophone in a typical case.*

cm^3, $C = 3 \times 10^{-5}$. In Fig. 8 the ordinates represent the ratio of the pressures calculated by formula (5) to those calculated by (4). The dotted curve represents the calculations from equation (5) with the correction (6) for wall conductivity omitted. In the case considered the pressures calculated from Wente's formula are about 22% lower than those given by (4) at 30 cycles.

9. Summary of Assumptions.—The more important assumptions underlying equation (4) are as follows:

(1) The walls of the enclosure are perfectly rigid.

(2) The linear dimensions of the enclosure are small compared with the wavelength of the sound.

(3) The "wavelength" of the diffusive thermal wave emanating from the strip is small compared with the distance between the strip and the wall.

(4) The average temperature can be calculated by the method represented by equation (6), Note 2.

(5) The loss of heat from the strip due to radiation and conductivity is negligible.

The following brief critical discussion of the effect of these assumptions and corrections for them may be of interest.

10. Effect of the Motion of the Diaphragm.—The assumption of perfect rigidity of the walls of the enclosure is chiefly violated by the motion of the diaphragm and by the flow of gas through the capillary ducts used for the supply of hydrogen.

To correct for these effects it seems proper to go back in the derivation of the pressure equation to equation (4), Note 2, and substitute for it the more general perfect-gas relation:

$$dp/P_0 = d\overline{T_1}/T_a - dV/V \tag{7}$$

where dV is the change of volume due to the lack of perfect rigidity. Expressing dV in terms of the mechanical impedance per unit area ($Z=R+iX$) of the yielding surface, of area S', we have

$$p_1 = \frac{P_0}{T_a}\,\frac{\overline{T_1}}{1 + P_0S'/i\omega Z}\,. \tag{8}$$

Substituting the value of T_1 previously employed and carrying through the subsequent work (equations 3 to 10, Note 2) we arrive finally at the following formula for the pressure:

$$p_1 = .48I_0I_1R/(\omega mCVA\alpha D^{1/2}/2S) \tag{9}$$

where

$$D = \left\{1 - \frac{4kS^2}{\omega CVA} - \frac{2SS'T_a}{\omega^2CVAmZ^2}\left[\frac{\omega C}{2S}(R+X) + 2k\alpha R\right]\right\}^2 + \left\{1 + \frac{4kS^2}{\omega CVA} + \frac{4S}{VA\alpha} + \frac{2SS'T_a}{\omega^2CVAmZ^2}\left[\frac{\omega C}{2S}(R-X) - 2k\alpha X\right]\right\}^2.$$

This formula replaces (4) and becomes identical with it when $Z=\infty$ or $S'=0$.

The impedance of the diaphragm varies with frequency in a rather complicated way; it will be sufficient to estimate the order of the effect at low frequencies when the impedance is approximately that of a pure compliance.

For a uniform steady pressure the average impedance of the diaphragm per unit area is:

$$Z = iX = -i8t/\omega R^2 \tag{10}$$

where t is the tension of the entire thickness of the membrane and R is the membrane radius.

In a typical case of a duralumin membrane, 4.2 cms in diameter, stretched to a natural frequency (undamped) of 5350 cycles, $t = 6 \times 10^6$ dynes/cm. If typical values for the physical constants of the thermophone are assumed the pressure calculated from (4) differs 5% from that calculated from the more accurate equation (9) in the case of air and by 2.5% in the case of H_2. We therefore conclude that the effect of the motion of the diaphragm will be negligible in most practical cases.

11. Leakage of Sound Through the Capillary Ducts.—Assume that there are two ducts for the supply of hydrogen, or for ventilation in the case of air, of length l and area S' and that these ducts are terminated at the remote end by a system of tubing of such large diameter as to offer a negligible impedance. The impedance per unit area of each capillary tube, looking from the enclosure, is approximately:

$$Z = R + iX = 7.1\gamma' l(f\rho)^{1/2}/d + i\omega\rho l \tag{11}$$

where

$$\gamma' = 1.54\mu^{1/2}$$

and μ is the viscosity of the gas. For air $\mu = 18.6 \times 10^{-5}$ poise and for H_2, $\mu = 9 \times 10^{-5}$ poise, so that

$$\begin{aligned} R(\text{air}) &= .0054 f^{1/2} l/d; \qquad X = .0082 lf \\ R(H_2) &= .001 f^{1/2} l/d; \qquad X = .00056 lf. \end{aligned} \tag{12}$$

From a physical viewpoint the inertia of the gas in the capillary tubes and the compliance of the gas in the thermophone enclosure constitute a simple Helmholtz resonator in which there is a possibility of resonance. If the resistance is small the condition for resonance is:

$$\omega^2 = 2\gamma P_0 S'/Vl\rho. \tag{13}$$

In two equal capillary tubes of $\frac{1}{4}$ mm diameter and 2 cm length and a thermophone volume of 10 cm^3 the resonant frequencies are 36 cycles for air and 264 for H_2. At these frequencies, however, equation (12) shows that the resistance is several times the reactance so that any rise of pressure due to resonance may be expected to be pretty well damped out.

Numerical calculations for a typical case are shown in Fig. 9 where the ordinates represent the ratio (multiplied by 100) of the true pressure from equation (9) to the pressure calculated from formula (4). The case considered is that of two capillary tubes $\frac{1}{4}$ mm diameter and

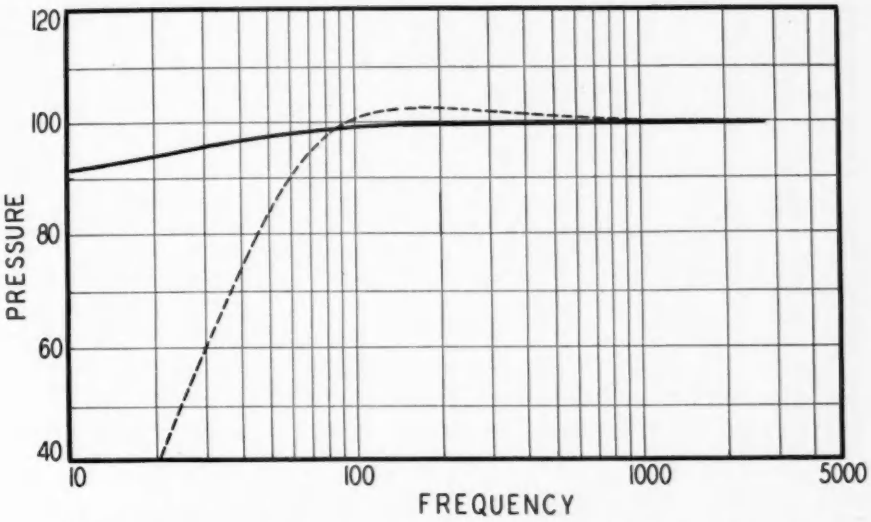


FIG. 9. *Loss of pressure (solid curve) due to leakage of sound through capillary ducts in a thermophone; 2 ducts, 1/4 mm bore, 2 cm long.*

each 2 cm in length, the gas being hydrogen. It may be concluded that for tubing of this diameter a length of 5 to 10 cms will be adequate for precise work.

As in the cases of the effects of wall conductivity and of the motion of the diaphragm an attempt at an independent "first-order correction" for the loss of sound in the capillary tubes leads to a correction which is too large. The improper correction factor obtained by this method is shown by the dotted curve in Fig. 9 for comparison.

12. Effect of the Dimensions of the Enclosure.—The upper limit of the frequency range over which the pressure formula is valid is mainly determined by the breakdown of the assumption of constant pressure over the enclosure which occurs when the wave-length of the sound becomes

comparable with the dimensions of the enclosure. The largest dimension is the diameter and in a circular cavity the characteristic frequencies for radial vibrations are given by the roots of $J_1(\omega R/c)=0$. For a diameter of 4 cm and air the first two characteristic frequencies are 10,000 and 18,300 cycles. The velocity of sound (c) in hydrogen is approximately 4 times that in air so that the corresponding frequencies in H_2 would be at 40,000 and 73,000 cycles. The advantage of hydrogen over air in extending the upper frequency limit was pointed out by Arnold and Crandall and this gas is now generally employed. It is somewhat less suitable than air at the lower frequencies due to the greater effects of wall conductivity and the greater "wavelength" (about 2.5 times)

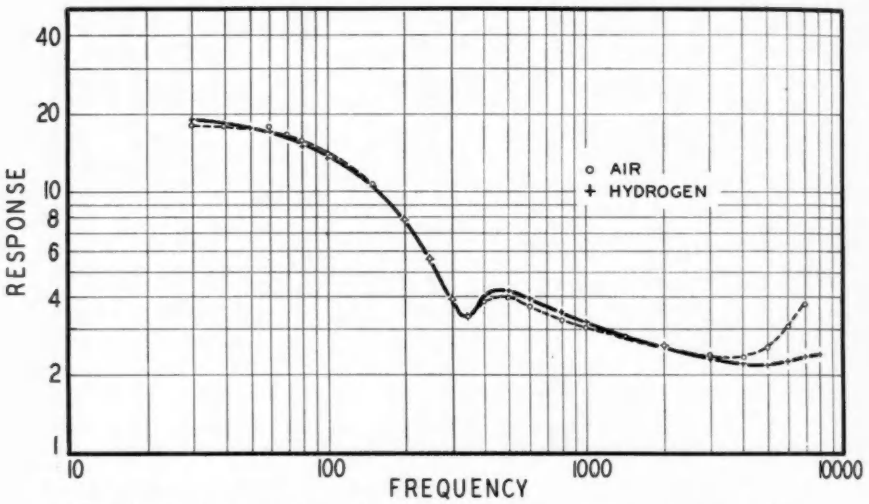


FIG. 10. *Comparison of thermophone calibrations with air and hydrogen showing spurious pressure rise with air at high frequencies.*

of the diffusive thermal wave, but this disadvantage is not usually serious in ordinary work. A comparison of the experimental results obtained at high frequencies in hydrogen and in air is shown in Fig. 10. As the frequency increases the pressure seems to build up near the middle of the diaphragm, thus increasing the response. In Fig. 10 this effect begins to be noticeable at 2,000 cycles and amounts to 60 percent at 7,000 cycles.

13. Other Assumptions.—Other assumptions, (3) and (4), were considered in the derivation of equation (4). Their effects are chiefly confined to low frequencies. The difficulty of the exact calculation of the temperature distribution at low frequencies is perhaps the outstanding

defect of the theory. For this reason it is desirable in the practical construction of the thermophone to approach as closely as possible the idealized one-dimensional problem actually contemplated. To this end the strips should be wide, close together and as extensive in their plane as possible. It may be noted that from the view point of these particular assumptions air is more suitable than hydrogen on account of its lower diffusivity.

14. Values of Some of the Physical Constants in the Thermophone Formula.—Most of the physical constants occuring in equation (4) are functions of the temperature. The best experimental data for some of these constants are recorded here for convenience of reference.

Consider first the thermal conductivity, k, of the gas. The reported experimental values of the conductivity of gases vary considerably. For example the 19 determinations of k for air cited by the *International Critical Tables* (I.C.T., Vol. V, p. 213) deviate on the average from the weighted mean by 7 percent. According to the I.C.T. the best experimental data for air and hydrogen may be represented by

$$\text{Air:}\quad k(T) = 5.34(1+.003T)\times 10^{-5}\ \text{cal/cm}^2\text{/sec}$$
$$\text{H}_2\text{:}\quad k(T) = 38(1+.0028T)\times 10^{-5}$$

where T is the temperature in degrees C. Wherever k occurs explicitly in equation (4) it is to be taken at the average temperature of the enclosure, which may be assumed to be $\frac{1}{2}(T_{s0}+T_w)$, where T_{s0} is the steady strip temperature due to the current and T_w is the temperature of the walls of the enclosure.

The factor $\alpha = (\pi f \rho C_p/k)^{1/2}$ involves ρ, C_p and k which are to be given their values at the temperature of the strip, T_{s0}. C_p varies but little with temperature and in the range $0-100°$C and normal pressure has the values (Landolt & Bornstein, 5th Ed., II, p. 1274):

$$C_p\ (\text{air}) = 0.24$$
$$C_p\ (\text{H}_2) = 3.41$$

The density has the following values at standard atmospheric pressure:

$$\rho(\text{air}) = 1.29(273/T)\times 10^{-3}$$
$$\rho(\text{H}_2) = .09(273/T)\times 10^{-3}$$

where T is the temperature of the gas in degrees absolute. We thus have for α:

$$\alpha(\text{air}) = 4.25 f^{1/2}[273/(273+T)(1+.003T)]^{1/2}$$
$$\alpha(\text{H}_2) = 1.59 f^{1/2}[273/(273+T)(1+.0028T)]^{1/2}$$

where T is the strip temperature in degrees C.

I have found it convenient to adopt a standard difference of temperature between strip and enclosure walls. This simplifies calculations and when a set of strips becomes accidently damaged the heating current is easily readjusted, if necessary, to obtain the standard temperature rise. The standard values which have been employed are 50°C for air and 10°C for hydrogen. Hence for a 20° ambient temperature, $T_{s0}=343°$ abs., $T_a=318°$ abs. for air; and $T_{s0}-303°$ abs., $T_a=298°$ abs. for hydrogen. At these temperatures several of the quantities in equation (4) assume the following values:

Air	Hydrogen
$\alpha=3.44f^{1/2}$	$\alpha=1.45f^{1/2}$
$m=10\times10^{-5}$	$m=8.8\times10^{-5}$
$A=2.23$	$A=2.38$
$k=6.45\times10^{-5}$	$k=41.3\times10^{-5}$.

The total heat capacity of the gold foil (mass x specific heat) may be determined by weighing the leaf from which the strips are to be cut and assuming that the specific heat has not been changed by the beating from the value for the bulk metal (a 9×9 cm leaf weighs about 12 milligrams). The specific heat of gold is 3.16×10^{-2} so that if M is the weight of the leaf per cm² then: $C=0.0316\ MS$.

It is also necessary to determine the relation between the heating current and the steady temperature T_{s0} assumed by the strip. This can be found from measurements of the resistance of the strip, the required accuracy may be obtained by means of a potentiometer. A standard resistance R_s, having approximately the same resistance as the thermophone strips, is used. The strip current I_0 is varied over an estimated range (say 0 to 0.5 ampere) and the ratio of the strip resistance to R_s noted for each I_0. This permits calculation of the ratio of the strip resistance with current flowing to the resistance with no current. If the temperature coefficient of resistance a be known, the temperature can be calculated. There is some doubt as to the propriety of employing the temperature coefficient obtained for massive gold, i.e. $a=0.003$, in the case of the foil. Bridgman[12] has reported measurements of a on two samples of gold leaf of different thicknesses which gave values of .0015 between 0° and 30°, or about one-half the value for massive gold. My own measurements have given values of .00134, .00142, .0013, .00148, .0017 for various specimens, which, combined with Bridgman's values give a mean of about .0015.

[12] P. W. Bridgman: *Proc. Amer. Acad.*, **57**, p. 145, April, 1922.

If the above measurements have been carried out in air and the current required to produce a given strip temperature in hydrogen is desired this can readily be computed as follows: Let ΔT_1 and ΔT_2 represent the difference in temperature between the strip and enclosure walls for air and H_2 respectively. Let I_{01} and I_{02} represent the corresponding steady currents. Then since ΔT is proportional to I_0^2/k

$$\frac{I_{01}^2}{I_{02}^2} = \frac{\Delta T_1}{\Delta T_2}\,\frac{k_1}{k_2}.$$

For a standard rise of 10° for H_2 and 50° for air

$$\frac{I_0(\text{Air})}{I_0(\text{Hydrogen})} = .835.$$

At high frequencies, when the amplitude of the a-c component of current I_1 may become large, it may be desirable to remember that the total steady heating effect is proportional to $I_0^2+I_1^2/2$, and to vary I_0 to keep this quantity and T_{s0} and T_a constant.

The relative numerical values of the terms in the denominator D of equation (4) for a thermophone having the physical characteristics $S=6.3$ cm², $V=8.5$ cm³, $C=3\times10^{-5}$ are as follows:

	Air	Hydrogen
(1)	$4kS\alpha/\omega c = 30/f^{1/2}$	$80/f^{1/2}$
(2)	$4kS^2/\omega CVA = 3.05/f$	$17.4/f$
(3)	$4S/VA\alpha = 0.39/f^{1/2}$	$0.85/f^{1/2}$

For work of ordinary engineering accuracy terms (2) and (3) may be ignored, and the following simplified formula may be recommended:

$$p_1 = \frac{0.153 I_0 I_1 RS}{fmCV\alpha A\sqrt{1+(1+4kS\alpha/\omega C)^2}}. \qquad (14)$$

The pressure developed by the thermophone comprises, in addition to the desired output of fundamental frequency, a component of double frequency. This is discriminated against to a large extent by the frequency characteristics of the thermophone and at medium and high frequencies the ratio of second harmonic to fundamental is approximately equal to $I_1/10I_0$.

It is also possible to omit the d-c current I_0 altogether. In this case the alternating pressure output is of double frequency and the factor 0.48 I_0 I_1 in (1) is replaced by 0.24 I_1^2 where I_1 is the peak-value of the alternating current. This method is not convenient in calculation since

T_{s0} and several of the quantities in (1) vary with I_1 which has to be changed to maintain a more or less constant pressure output with frequency.

15. Effect of the Microphone Polarizing Voltage Upon the Response.– The electric field between the back-plate and diaphragm which is produced by the polarizing voltage exerts a mechanical force of attraction:

$$F = -\frac{E_0^2}{2}\frac{\partial C}{\partial x} \tag{15}$$

upon unit area of the diaphragm. Here C is the capacity between the back-plate and diaphragm for the unit area and x is the displacement of the diaphragm. This force is of the nature of that produced by a negative stiffness and therefore acts to reduce the affective stiffness of the diaphragm. Let the stiffness due to the electric field be called the *electric stiffness* and denoted by S_e.

The total stiffness of the diaphragm in terms of the deflection at the center for uniform unit force is $S = S_d + S_e$, where

$$S_d = 4T/R^2 \tag{16}$$

$$S_e = 4.45 \times 10^{-7}\frac{E_0^2 R_B^2}{R^2}\left(2 - \frac{R_B^2}{R^2}\right) \quad \text{(volts, cms)}$$

approximately. R_B and R are the radii of the back-plate and membrane respectively and T is the membrane tension.

An idea of the relative values of these quantities in practical apparatus may be obtained from the following calculations. Let $R_B^2/R^2 = .85$, $R = 2.15$ cm; for a duralumin diaphragm .0012″ thick stretched to $f. = 5{,}000$ cycles, $T = 7 \times 10^6$ dynes/cm so that if $d = .001''$

$$S_d = 6 \times 10^6$$
$$S_e = 1.12 \times 10^6$$

for $E_0 = 200$ volts. The low frequency response of the microphone is roughly proportional to $1/S$ and in the above case would be increased about twenty percent by the negative compliance due to the polarizing voltage. The effect would be less at high frequencies so that the *shape* of the frequency response characteristic would be somewhat modified. Obviously in such cases a statement of the response should include the value of polarizing voltage.

16. Pistonphone.—The pistonphone is a more direct means of applying alternating air pressures to the diaphragm than either of the foregoing methods. It comprises a chamber, of which one wall is formed by

the diaphragm, and of which part of the remaining boundary is formed by a piston or diaphragm which is vibrated by means of a reciprocating mechanical movement operated by a motor. Frequencies up to about 200 cycles may be obtained. On account of the simplicity of the pressure calculation the device is useful in checking the absolute values of calibrations performed by other methods at low frequencies. In particular it supplements the thermophone over a range of frequencies in which the thermophone theory is weakest. The pistonphone has been extensively employed by Wente and for details of suitable mechanical constructions the reader is referred to his papers (i.e. Reference 11).

The walls of the pistonphone enclosure are generally metallic, or at least partly so, and it is necessary to consider their cooling effect in calculating the generated pressure. The following formula for the pressure developed is derived in Note 3:

$$p_1 = \frac{\gamma P_0 S_p \zeta_1}{V}\left\{\frac{1}{1+\frac{(\gamma-1)S_w}{\alpha V}+\frac{1}{2}\left(\frac{(\gamma-1)S_w}{\alpha V}\right)^2}\right\}^{1/2} \tag{17}$$

where

P_0 = ambient pressure
γ = ratio of specific heats for air (=1.4)
$\alpha = (\omega\rho C_p/2k)^{1/2} = 3.7f^{1/2}$ for air
S_w = area of metallic walls
V = volume of the enclosure
S_p = effective area of the piston
ζ_1 = amplitude of motion of the piston.

The bracketed term may be regarded as a correction for the departure from a purely adiabatic compression which is caused by the thermal conductivity of the walls.

This formula differs somewhat from that given by Wente (reference 11, p. 243), which may be written in our notation:

$$p_1 = \frac{\gamma P_0 S_p \zeta_1}{V}\left\{1-\frac{S_w}{\alpha V}+\frac{S_w^2}{2\alpha^2 V^2}\right\}^{1/2}. \tag{18}$$

The corrections given by these formulae in a practical case are tabulated below. The following numerical data have been assumed:

$$S_w/V = 1.5; \qquad \alpha = 3.7f^{1/2}.$$

A small effect is predicted by either formula and in most cases will be trivial in comparison with the other sources of error.

TABLE I

COMPARISON OF THE CORRECTIONS FOR WALL CONDUCTIVITY FOR THE PISTONPHONE GIVEN BY EQUATION (17) AND WENTE'S FORMULA. VALUES REPRESENT PERCENTAGE REDUCTION OF GENERATED PRESSURE

Frequency (Cycles)	Equation (17)	Wente's Formula
10	2.5%	6%
20	1.8	4.5
30	1.5	3.7
60	1.0	2.5
100	0.8	2.0
200	0.5	1.42

II. WAVE RESPONSE

17. Method Employing the Rayleigh Disk.—Calibration in terms of the pressure in a plane progressive sound wave normally incident upon the face of the microphone may be performed by using the Rayleigh disk to measure the velocity v in the wave field. In a progressive sinusoidal plane wave the pressure is simply related to the velocity by:

$$p = \rho v c = 42.6v \text{ (for air)}. \tag{19}$$

The Rayleigh disk is extremely delicate and on account of the necessity of protecting it from draughts measurements must be made in a closed chamber. By proper treatment of the walls the reflection in the chamber may be greatly reduced, nevertheless it is difficult to secure adequate absence of standing waves at the lower frequencies, say below 1000 cycles, in a room of ordinary size ($25' \times 25' \times 14'$). Fortunately, frequencies below this are not important since the pressure and wave calibrations tend to approach equality as the frequency is diminished; in microphones of conventional size they are usually within about 10% of equality at 1,000 cycles. To obtain the best possible approximation to a free wave field the distance between microphone and source should be small in comparison to the linear dimensions of the chamber. In order to obtain the best approximation to a plane wave the source should be small in comparison with the distance between it and the microphone. A (small) source without a baffle is also desirable to avoid standing waves between microphone and source.

An arrangement of apparatus for a normal wave calibration employed by the writer is illustrated in Fig. 11a. The plan is not to scale but the principal dimensions are marked. The source of sound is a moving-coil

electrophone unit terminated by a tube whose length can be adjusted for resonance. The diameter of the tube is about $\frac{1}{2}''$. The rotation of the disk is observed optically by means of a lamp and scale. Observations are made outside the chamber by means of a telescope mounted in the door.

To avoid the necessity for opening the door during a frequency run the source shown is sometimes replaced by a small horn 2″ in diameter, driven by a moving-coil and specially damped so as to give a

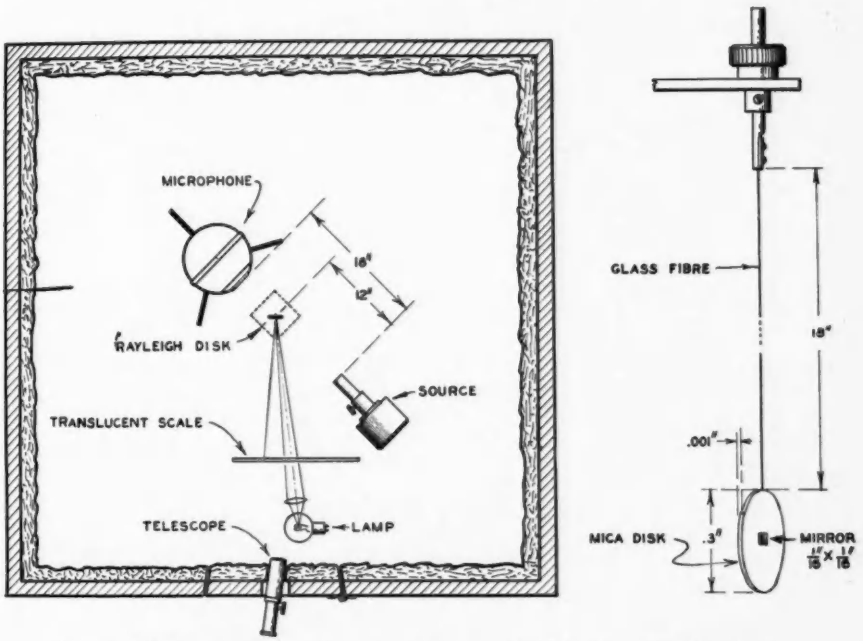


FIG. 11. (*a*) *Arrangement of apparatus for wave calibration by means of the Rayleigh disk;* (*b*) *dimensions of typical Rayleigh disk system.*

substantially uniform output over the frequency range employed. Errors arising from the proximity of the Rayleigh disk to a source of finite size have been discussed by West.[13]

18. Rayleigh Disk.—This is a light circular disk suspended by means of a delicate fibre so as to rotate about a vertical axis against the restoring torque of the fibre. If a flat object of this sort be suspended in the path of a current of air it tends to turn itself so that its plane is normal to the direction of the air stream. This tendency persists if the direction of air flow be reversed so that, as first suggested by Lord

[13] W. West: *Jour. Inst. Elec. Eng.*, **67**, 1140, September 1929.

Rayleigh,[14] a steady deflection will be obtained in a periodic air current and the device may be used to measure the particle velocity in a sound wave.

The steady torque so produced by a circular disk in an inviscid fluid of sinusoidally varying velocity has been calculated by Koenig. For a disk of negligible thickness:

$$\text{Torque} = \frac{1}{6}\rho d^3 v^2 \sin 2\theta \tag{20}$$

where ρ = density of the gas
d = diameter of disk
v = rms velocity of gas
θ = angle between the direction of velocity and the normal to the disk.

Maximum sensitivity is obtained when $\theta = 45$ and the disk is generally so adjusted. The steady deflection against the torque of the fibre is thus roughly proportional to v^2.

References on the Rayleigh disk technique are given below.[15] An exceptionally valuable article on the manipulative aspects of the subject has been published by E. J. Barnes and W. West (*Jour. Inst. Elec. Engs.*, 65, 871, September, 1927). The following practical suggestions on the construction and use of the disk may be of interest.

From the viewpoint of sensitivity the disk should be as large as possible since the torque increases as the cube of the diameter. Koenig's equation is not valid, however, unless the diameter is small compared with the wavelength of the sound, so that the increase in diameter must be carried out judiciously if accuracy at the higher frequencies is desired. Some guidance in this respect is furnished by experiments performed by Barnes and West who found a definite error when the ratio d/λ reached 0.13. If this is taken as the limiting error a diameter of less than 1 cm is indicated at 10,000 cycles.

The dimensions of a satisfactory disk and suspension are shown in Fig. 11b. The disk is of mica 0.001″ thick and 5/16″ diameter. A small mirror, 1/16″, is attached to the face. This is cut from a No. 1 micro cover glass which has been silvered. If the mica disk is not too far from plane it is sometimes possible to silver the disk and so dispense with a cemented mirror. The suspension is a soft glass thread and is attached to

[14] Rayleigh: *Phil. Mag.*, **14,** 186, 1882.

[15] Mallett and Dutton: *Jour. Inst. Elec. Eng.*, **63,** 502, 1925. Geiger and Scheel: "Handbuch der Physik," Vol. 8, Akustik, p. 573 (Berlin: 1927).

the disk by beeswax or shellac. Glass fibres of sufficient fineness are readily made by melting glass tubing in a blowpipe flame, drawing out one end, attaching it to a rotatable drum 3 feet in diameter and winding up the glass thread at the proper rate. Black velvet is useful as a back ground in handling such fibres. The moment of torsion S of the suspension may be measured by attaching a rather larger and heavier disk than the one to be employed (say 3 cms in diameter) and measuring the free period of rotatory oscillation. If M is the mass of the disk in gms, T the period in seconds, d the diameter in cm, then:

$$S = Md^2\pi^2/4T^2. \tag{21}$$

The large disk is then removed and the disk to be used is fastened to the fibre. Since S is known v can now be calculated from the observed deflection by Koenig's formula:

$$v(\text{rms}) = 2.43 \times 10^{-3} \sqrt{S\phi/d^3} \tag{22}$$

where ϕ is the angular deflection in radians. It is assumed that when $\phi = 0$ the disk is 45° to the wave normal.

Anomalous results are often caused by resonant vibrations of the disk. The effect ordinarily extends over a small frequency range (say 50–100 cycles) and may give rise to errors of 10 to 15 percent (West). These effects are easily identified by exploring about a suspected frequency point.

One of the most troublesome manipulative features of the Rayleigh disk is its sensitivity to draughts. This difficulty may be greatly alleviated by enclosing the disk in a screen made of open-mesh silk or other fabric. A high quality closely woven cheese cloth such as used for dental napkins is quite suitable. The loss of sound through such material is almost immeasurable and certainly less than 5% up to 10,000 cycles.

In this sort of work the disk should be completely open and not encumbered with the tubes which are often employed to increase its sensitivity (by resonance) and to protect it from draughts. The distortion of the sound field by such obstacles may introduce important errors at high frequencies.

19. Increase of Pressure by Diffraction.—If the microphone diaphragm were of infinite extent, or part of an infinite wall, the pressure at the diaphragm due to an incident sound wave would be doubled at all frequencies. If, on the other hand, the dimensions of the microphone were small in comparison with the wavelength of the sound, the pressure at any point of its surface would tend to approach that in the undisturbed

sound field. In the actual case, in which neither of these conditions obtains, the pressure increase due to diffraction varies from unity at low frequencies to 2 at high frequencies.

The nature of this variation may be brought out by considering the case of a spherical obstacle for which the increase of pressure can be calculated theoretically. The pressure ratio for a sphere of diameter a and a plane progressive wave of the type $Exp\ iw(t-r/c)$ may be expressed in terms of Bessel functions whose order is half an odd integer as follows:

$$\frac{P}{P_0} = \sqrt{\frac{2}{\pi ka}} e^{-ika} \sum_{n=0}^{\infty} \frac{-(2n+1)i^n P_n(\cos\phi)}{(-)^n(nJ_{-n-1/2} + kaJ_{-n-3/2}) - i(nJ_{n+1/2} - kaJ_{n+3/2})} \tag{23}$$

where the arguments of the Bessel functions are ka, $k=2\pi f/c$, c is the velocity of sound, and ϕ is the angle between the wave-normal and the axis of the sphere. Numerical calculations for $\phi=0$ over a large range of ka have been performed and tabulated in an earlier paper (Reference 1). The pressure ratios for spheres of various diameters are shown in Fig. 12, as a function of the frequency.

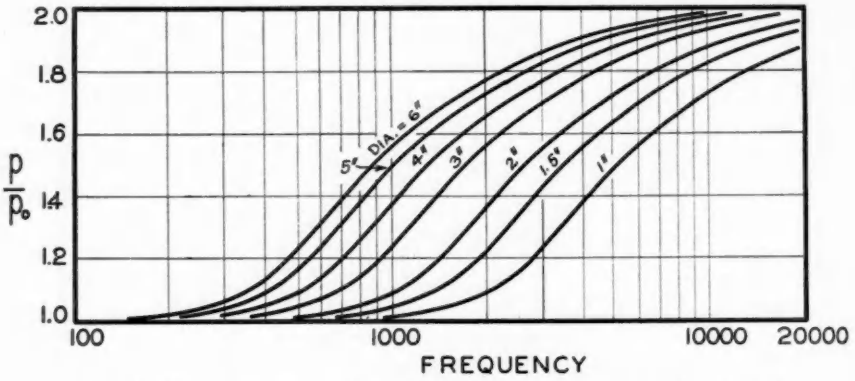


FIG. 12. *Increase of pressure due to diffraction for a plane wave and spherical microphone housings of various diameters.*

In order to render the diffraction effect definite and calculable the microphone may be built into a spherical housing. The microphone may be designed so that the diaphragm is substantially a part of the spherical surface. In this way cavity resonance is eliminated and the pressure ratio is due only to diffraction. A standard microphone of this type is shown in Fig. 13.

The pressure-ratios as experimentally measured by means of the

Rayleigh disk for a group of microphones (Fig. 13) are shown in Fig. 14. The curve for the spherical microphone agrees pretty well with the

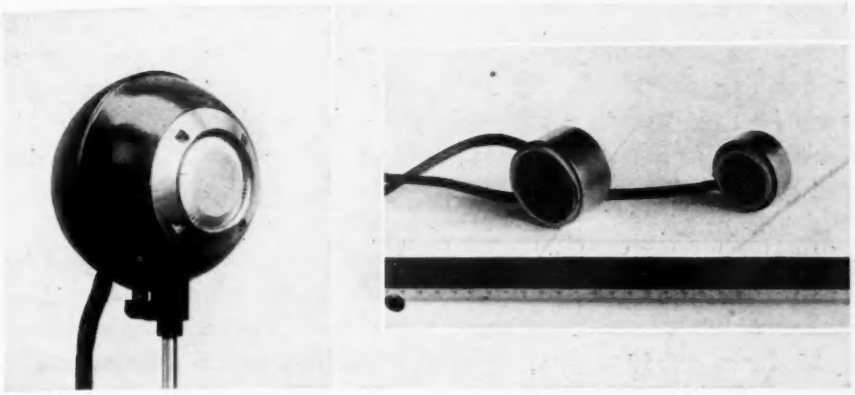


FIG. 13. *Group of microphones with flush diaphragms; (a) standard spherical condenser microphone; (b) and (c) electromagnetic microphones of small size.*

theoretical curve (dotted). The solid angle subtended by the diaphragm was approximately 35° and the diameter of the spherical mounting was

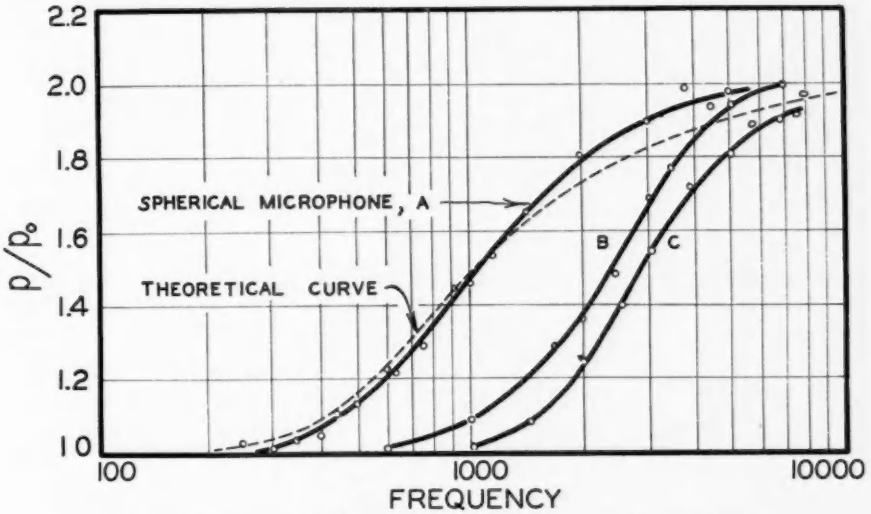


FIG. 14. *Experimental values of pressure ratios for group of microphones illustrated in Fig. 13.*

5 inches. The microphones shown at the right of Fig. 14 (B and C) are of the electromagnetic type and are comparatively small in size, the

diameter of C being only 1.6″. The diaphragms were in all cases of the flush type so that the pressure ratio is presumably a purely diffractive effect.

With a condenser microphone having a flush type diaphragm it is a comparatively simple matter to adjust the damping so that the response as given by the pressure calibration is complementary to the pressure ratio and thus to obtain a free wave response characteristic which is uniform over a large range of frequency. This has been carried out in the design of the standard spherical microphone, A Fig. 13, which is intended for automatic recording of the frequency response characteristics of loud-speakers. This result may also be obtained by equalization in the electrical circuits since the pressure ratio curve is of a type which is easily equalized.

20. Increase of Pressure by Cavity Resonance.—In microphones of conventional form the diaphragm is situated at the bottom of a shallow cavity and an increase of pressure is produced by resonance in this cavity. The effect may be calculated roughly if it is assumed that the cavity is provided with an infinite flange and that the velocity is uniform over the orifice. Such calculations have been given in an earlier paper,[16] and by West[17] who took into consideration the effect of the diaphragm impedance. If the impedance of the diaphragm is high, as is usually its case, the pressure ratio may be expressed as follows:

$$\frac{p}{p_0} = \frac{2}{i \sin kl[R_1 + i(X_1 - \cot kl)]} \tag{24}$$

where $k = w/c$, l is the depth of the cavity and

$$R_1 = 1 - J_1(2kR)/kR$$
$$X_1 = K_1/2(kR)^2$$

where R is the radius of the diaphragm, ρc is the characteristic impedance for air ($=42.6$) and J_1 and K_1 are Bessel functions. The factor 2 occurs in (24) because the pressure is doubled by the infinite flange.

If it be assumed that the effects of diffraction and cavity resonance operate independently the total pressure ratio is the product of (23) and (24) divided by 2.

Comparisons of the effect as calculated in this way with the experimental values obtained by means of the Rayleigh disk have been made

[16] Ballantine: Reference 2, p. 3.
[17] West: Reference 5, p. 7.

and are in rather better agreement than we have a right to expect in view of the nature of the underlying assumptions.

Pressure-ratio curves as determined from the pressure and wave calibrations for two specimen microphones are shown in Fig. 15. The contours and dimensions of the microphones are given in the lower part of the figure. Microphone A was of the spherical type with recessed di-

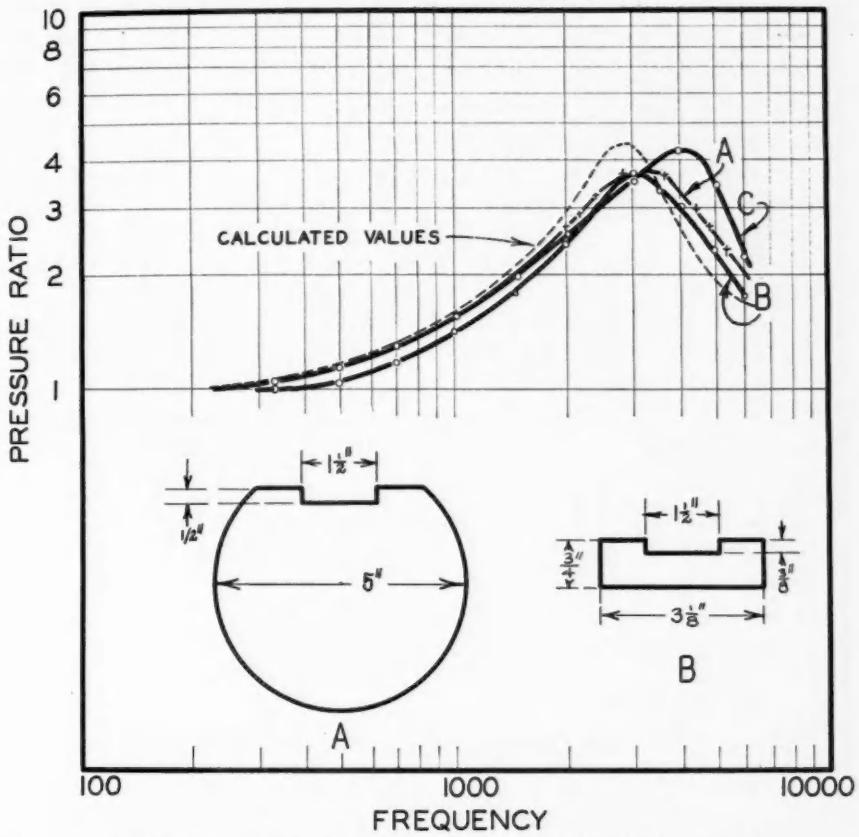


FIG. 15. *Pressure ratios for several condenser microphones having recessed diaphragms showing effects of cavity resonance.*

aphragm. The dotted curve represents the theoretical values calculated from (23) and (24). The rise of pressure is seen to amount to about 4×, or 300 percent. Microphone B was of more conventional form and was measured without the diaphragm protecting grille and amplifier housing. The dimensions of microphone B were about the same as those of several types sold commercially, and the pressure curve shown at B, Fig. 15, is in good agreement with that published by West for a Western

Electric Type 370W microphone. This form of curve may be used as an approximate correction for microphones of such contour when a Rayleigh disk calibration is not available, or as a curve by which old investigations may be corrected. In making these measurements the protecting grille for the diaphragm was absent and the customary amplifier stage and microphone mounting were omitted; hence the results are not strictly applicable to actual mountings of commercial instruments.

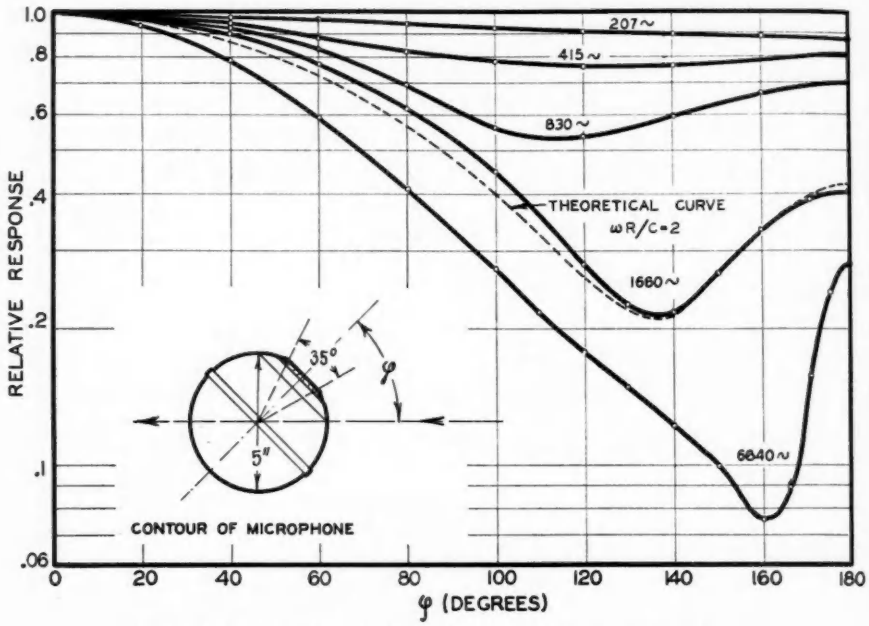


FIG. 16. *Directivity characteristic (plane waves) of spherical condenser microphone (a, Fig. 13).*

21. Directivity.—The directivity may be determined by rotating the microphone in a plane progressive wave field. An approximation to such a field may be obtained by placing the microphone at a sufficient distance from a small source, or at a short distance from an extended plane source, such as a large electrostatic loud-speaker. If the wave be sufficiently plane no correction is required for the variation of intensity with distance; otherwise a correction is necessary, since the rotation of the microphone changes the distance of the diaphragm from the source. The work should be carried out in the open air in view of the sensitivity of the microphone to reflection from the walls of a sound chamber when it is turned so that the angle between its axis and the wave normal is large.

The directivity curve for the spherical microphone (Fig. 13) is shown in Fig. 16. The frequencies were chosen to correspond to particular values of wR/C. The values for equal positive and negative values of ϕ differed slightly and mean values have been plotted.

The dotted curve for 1660 cycles represents values computed from the calculations made by Lord Rayleigh[19] for the pressure rise at a *point* in the surface of a sphere. The membrane subtended a solid angle of 35° and Rayleigh's point values were integrated vectorially to obtain the average value over the diaphragm. The experimental values are in fair agreement with the values calculated in this manner.

The smoothness of the directivity curve for the spherical microphone does not fully come up to expectations. In spite of the fact that the geometrical figure of the mounting is of the simplest possible type and that the diaphragm is substantially a part of the spherical surface and all cavities have been eliminated some irregularities are observed at the higher frequencies. These may perhaps be attributable to a departure of the diaphragm from a piston-like action. Nevertheless this type of microphone offers some advantage in this respect over the conventional types.

22. Significance of the Pressure and Wave Calibrations for Various Uses of the Microphone.—A question may arise as to the significance of these calibrations for ordinary uses of the microphone. These may be divided into two classes: (1) pickup or sound measurement in an open sound field, and (2) measurement of pressure when the diaphragm is part of an enclosed space.

Examples of the latter class are telephone receiver testing by means of an artificial ear incorporating the diaphragm of the condenser microphone;[20] and the use of the microphone as a sonic probe.[21] Obviously in both cases the pressure response is of sole interest.

In the first class the uses fall into two groups; (1) the microphone may be employed to measure the sound pressure at some point of a sound field, or (2) to pick-up a sound for electrical transmission or recording.

It is obvious, from the directivity characteristic, that the microphone cannot be employed to measure the pressure at a point of any sound field which comprises a number of waves arriving from different direc-

[19] Rayleigh: *Scientific Papers*, Vol. 5, p. 151, 1904.

[20] Cohen, Aldridge and West: *Jour. Inst. Elec. Engs.*, p. 846, June, 1926. A. J. Aldridge: *Inst. Post Office Elec. Engs.*, No. 124, p. 41, 1929.

[21] Wente and Bedell: *Bell Tech. Jour.*, p. 1, January, 1928.

tions. A possible exception is the trivial case where the frequency is so low that the microphone directivity is nil. Another possible exception is the case of a highly reverberant environment, such as the large testing chamber at Riverbank, which may be assumed to be a sort of acoustical black-body hohlraum in which the waves reach the microphone in uniform quantities from all directions. In this case the response at any frequency may be obtained by averaging over the directivity curve as recently proposed by Sivian.[22]

In environs of ordinary reverberation the response will probably be equal to the point field pressure at low frequencies, and appropriate to the direct wave from the source at high frequencies. Three factors conspire to the latter result: (1) the directivity of the microphone; (2) the directivity of the source; (3) decreased reflection from the majority of reflecting surfaces at high frequencies. It is assumed, of course, that the source faces the microphone in the normal way. The effect of all three factors is to reduce the response to the reflected waves in relation to that due to the direct wave.

To what extent this is true under ordinary conditions is indicated by the following simple experiments. The source of sound was a cone-type coil-driven electrophone of somewhat less (6″) than the usual diameter. Records of the sound-pressure frequency characteristic of this source were made under conditions by means of an automatic optical recorder. A record (not reproduced) was first taken outdoors at a height of 24 feet to get the basic response in the absence of all reflections. A 5″ spherical microphone, shown in Fig. 13, was used. The upper left hand records (Fig. 17) were made in a small sound chamber (25′×25′×12′) heavily padded with balsa wool and hair felt. The upper curve was taken at a distance of 18″ from the source and, except for a few serrations at the low frequencies due to reflection, is substantially identical with that obtained outdoors. This may be taken as the true response curve of the source. At a distance of 8 feet, however, there is a considerable contribution by reflection at the lower frequencies but above about 1700 cycles the two curves are practically identical. The third curve ($\phi = 180°$) was taken at the same distance (8 ft.) but with the microphone turned 180° so as to face away from the source. With the exception of a small shift in frequency, due to the variation in position of the diaphragm caused by this rotation, the low frequency curves are closely alike. The response at higher frequencies differs somewhat, but not a great deal, from that predicted by the directivity characteristic.

[22] L. J. Sivian: *Bell Tech. Jour.*, p. 108, January 1931.

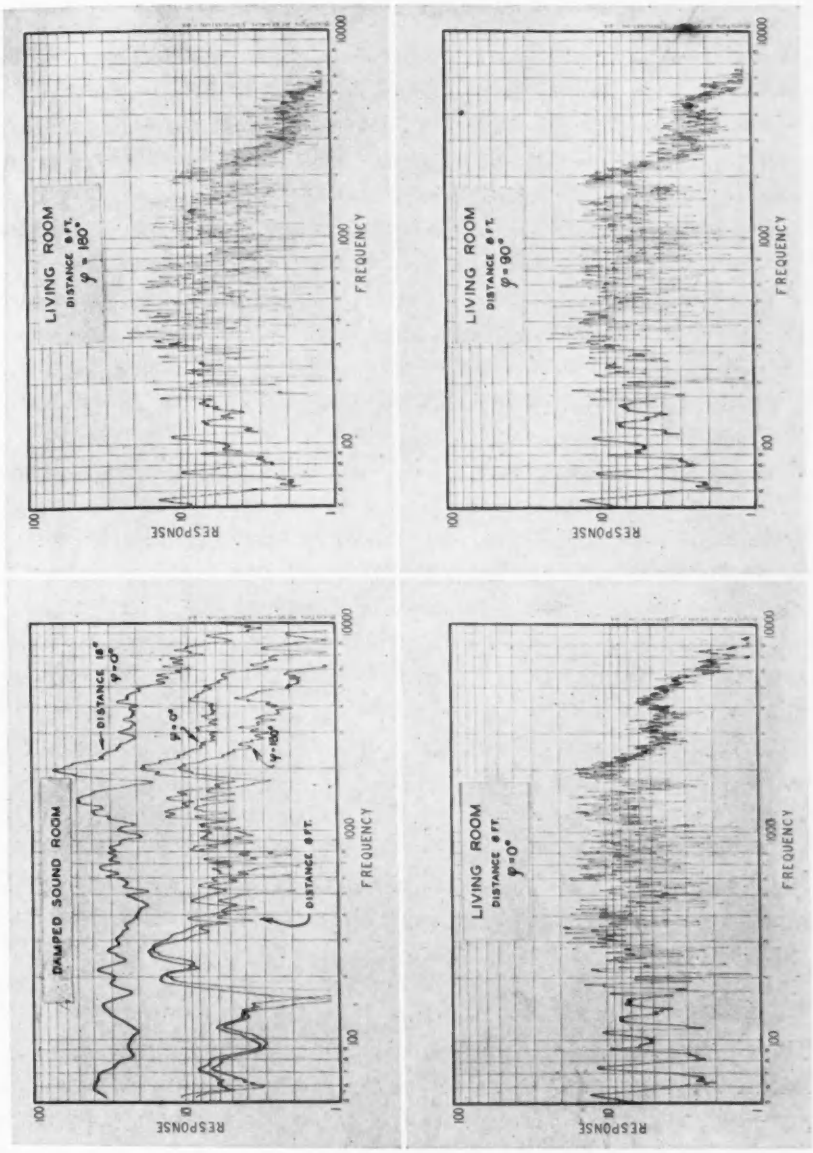


FIG. 17. *Sound-pressure-frequency records of a 6" piston taken in a damped sound room and in a room of average reverberation.*

The remaining three records were taken in a large room which had, for routine loud-speaker testing purposes, been so treated as to have about the damping of the average "living room" of the American home. The distance was the same as before (8 ft.) and the microphone was set at three different angles (0°, 90°, 180°) with respect to the direction of the source. The substantial similarity of the low frequency response for all angles is again apparent. In spite of the considerably greater part played here by reflection the space is far from being an acoustical "hohlraum," and for practical purposes the response is fairly predicted by the wave calibration and directivity characteristic.

A source of smaller directivity, such as the human mouth, may be expected to show greater reflection effects at higher frequencies but even in the extreme case it is doubted that any condition approaching "hohbaum" radiation would be experienced unless the reverberation was abnormally high.

A really doubtful case arises when the microphone is very close to the source, as in the case of telephonic pickup or in studies of human speech. This case requires further investigation. Calculations of diffraction in the case of a *spherical* wave show that the pressure-doubling tends to be more complete at any frequency as the microphone approaches the point source. At distances between speaker and microphone of the order of 1 foot, however, it is probable that the wave calibration will apply with fair approximation. It is useless to insist upon too much precision when the distance is small because of the uncertainties connected with reaction on the source.

Pending a more decisive inquiry it seems reasonable to conclude that for the usual technical applications of the microphone, the response may be predicted well enough for ordinary engineering purposes by the plane wave calibration and directivity characteristic.

NOTE 1

Calculation of the Mechanical Force Produced by the Grille

The electrostatic problem presented by the grille and diaphragm is illustrated in Fig. 2, and in Fig. A below. We require an expression for the average mechanical force per unit area over the diaphragm in terms of its dimensions.

This problem is most simply treated by the method of conjugate functions and the Christoffel-Schwarz transformation.[1] By means of

[1] For the theory of this transformation see any standard work on the function theory, e.g., Hurwitz-Courant "Funktionentheorie," 2nd Ed., p. 394, (Berlin, 1925).

this theorem a polygon in the z-plane representing the contour of the given system of conductors can be transformed into the real axis of a w-plane. If the solution of the w-plane problem so obtained is not obvious, a second transformation to a t-plane may be attempted in order to obtain a tractable problem (e.g. a linear field).

Let $a_1, a_2 \cdots a_n$ represent the values of w corresponding to the corners of the polygon in the z-plane, and let $\theta_1, \theta_2 \cdots \theta_n$ represent the internal angles of this polygon; then the required transformation is given by:

$$\frac{dz}{dw} = C(w - a_1)^{(\theta_1/\pi)-1}\ (w - a_2)^{(\theta_2/\pi)-1} \cdots (w - a_n)^{(\theta_n/\pi)-1}. \tag{1}$$

Our problem may be considerably simplified and the use of elliptic functions avoided if the breadth W of the bar is not too small compared with the distance d from the diaphragm. In this case C and D can be placed at $-\infty$ and the corners of the polygon $A\ B\ C\ D\ E\ F$ transformed into points $-\infty$, -1, -0, $+0$, a, $+\infty$ on the real axis of the w-plane respectively, as shown in Fig. A. Application of the theorem (1) gives the required transformation:

$$z = C_1 \int \frac{\sqrt{w+1}\, dw}{w\sqrt{w-a}} + D_1 \tag{2}$$

$$= C_1 \left\{ -\frac{2}{\sqrt{a}} \operatorname{Tan}^{-1} \sqrt{a} \sqrt{\frac{w+1}{w-a}} + 2 \operatorname{Tanh}^{-1} \sqrt{\frac{w+1}{w-a}} \right\}. \tag{3}$$

Let the constant D_1 be determined so that the zero of the z-plane, E, corresponds to the point $w=a$. The constants C_1 and a may be determined from the discontinuities which occur when w passes through the singularities (0 and ∞). To find C_1 let w approach $+0$ from $+a$. In this way we find:

$$C_1 = -id\sqrt{a}/\pi \tag{4}$$

a can be determined by letting $w=a$ when we should have $z=o$, from which:

$$\sqrt{a} = -S/2d. \tag{5}$$

Making these substitutions in (3):

$$z = \frac{i2d}{\pi} \left\{ \operatorname{Tan}^{-1} \sqrt{a} \sqrt{\frac{w+1}{w-a}} - \sqrt{a} \operatorname{Tanh}^{-1} \sqrt{\frac{w+1}{w-a}} \right\} - \frac{S}{2} + id. \tag{6}$$

The electrostatic problem in the w-plane is that of two plane conductors, one extending from $w=-\infty$ to $w=-0$, the other from $w=+0$ to $w=a$. This problem is intractable in its present form; a second transformation is therefore made to convert it into that of infinite parallel conductors in the t-plane, as shown in Fig. A. The required transforma-

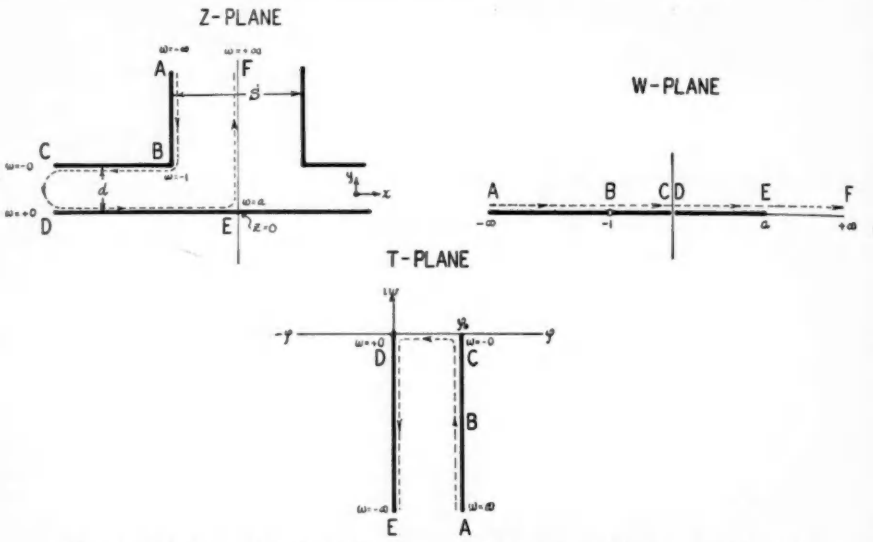


FIG. A. *Relating to the calculation of geometrical constant K for electrostatic grille.*

tion is obtained by applying (1) to the polygon $A\ C\ D\ E$ in the t-plane, as follows:

$$\frac{dt}{dw}=\frac{C_2}{w\sqrt{w-a}}+D_2 \tag{7}$$

so that

$$t=\frac{2C_2}{\sqrt{a}}\operatorname{Tan}^{-1}\sqrt{\frac{w-a}{a}}+D_2. \tag{8}$$

Let $D_2=0$ so that at the point D, $t=0$. Let $t=\phi+i\psi$, regarding ϕ as the potential function and ψ as the flux function. The potential of the conductor ED is taken as zero and that of the conductor ABC as ϕ_0. Then C_2 can be determined from the discontinuities in $t(=\phi_0)$ as w passes through zero:

$$t=-\frac{2\phi_0}{\pi}\operatorname{Tan}^{-1}\sqrt{\frac{w-a}{a}}. \tag{9}$$

The electric force E is equal to the modulus of the double transformation, that is:

$$E = \left|\frac{dt}{dz}\right| = \left|\frac{dt}{dw}\cdot\frac{dw}{dz}\right| \tag{10}$$

and from (2) and (7)

$$E = \left|\frac{C_2}{C_1\sqrt{w+1}}\right| . \tag{11}$$

Substituting the values of C_1, C_2, and from (9) the value of w in terms of t we get E in terms of t. At the surface of the diaphragm $E=\partial\phi/\partial y=\partial\psi/\partial x$. Letting $\phi=0$:

$$E = -\frac{\phi_0}{d}\frac{\cosh\psi}{\sqrt{a+\cosh^2\psi}} \tag{12}$$

where ψ is an abbreviation for $\pi\psi/2\phi_0$. The distance along the surface of the diaphragm x from the median plane EF is found by inverting (9), substituting the value of w so obtained in (6), and letting $\phi=0$, then:

$$x = \frac{2d}{\pi}\operatorname{Tanh}^{-1}\left(\frac{\sinh\psi}{\sqrt{a+\cosh^2\psi}}\right) - \frac{S}{\pi}\operatorname{Tan}^{-1}\left(\frac{\sqrt{a}\sinh\psi}{\sqrt{a+\cosh^2\psi}}\right) \tag{13}$$

remembering that for $x>1$, $\operatorname{Tanh}^{-1}x=\coth^{-1}x+i\pi/2$.

Let F represent the mechanical force on the electricity on the diaphragm per unit area. Then $F=\sigma E/2=E^2/8\pi$. Unfortunately our formulation expresses E in terms of ψ, not of x. Attempts at inversion lead to equations which are tedius to handle, so that the calculation is somewhat indirect. The average mechanical force F on the diaphragm between the median plane EF and a point on the diaphragm distant x from this plane is given by:

$$\begin{aligned}\overline{F} &= \frac{\overline{E}^2}{8\pi} = \frac{1}{8\pi x}\int_0^x E^2\cdot dx = \frac{1}{8\pi x(\psi)}\int_0^{\psi(x)} E^2(\psi)\frac{\partial x}{\partial\phi}d\psi \\ &= \frac{1}{8\pi x(\psi)}\int_0^{\psi} E\cdot d\psi .\end{aligned} \tag{14}$$

Performing the integration and substituting in the denominator the value of x from (13) we have finally:

$$F = \frac{\phi_0^2}{16\pi d^2}\cdot\frac{\log\left[1+\dfrac{2\sinh\psi(\sqrt{a+\cosh^2\psi}+\sinh\psi)}{1+a}\right]}{\operatorname{Tanh}^{-1}\left(\dfrac{\sinh\psi}{\sqrt{a+\cos^2\psi}}\right)+\dfrac{S}{2d}\operatorname{Tan}^{-1}\left(\dfrac{S^2}{4d^2}\dfrac{\sinh\psi}{\sqrt{a+\cos^2\psi}}\right)} \tag{15}$$

as an expression for the average force between point E and point x along the diaphragm surface. If x is now chosen as the point directly under the middle of a bar of the grille, that is $x=(W+S)/2$ (15) expresses approximately the average force per unit area produced by the grille. It is convenient to express this in the form:

$$\overline{F} = \frac{\phi_0^2}{8\pi d^2} K(x). \tag{16}$$

$\phi_0^2/8\pi d^2$ represents the force which would be produced by a plane mperforate electrode and K represents a function, calculable from (15), expressing the effect of the perforations of the grille.

Three variables are involved, S, W, and d. Ordinarily S and W are fixed by the grille construction and it is desirable to permit some variation in d. It is convenient to express K as a function of the universal arguments W/d and W/S, thus:

$$F = F_0 K\left(\frac{W}{d}, \frac{W}{S}\right). \tag{17}$$

Since $F=f_1(\pi\psi/2\phi_0)$ and $x=f_2(\pi\psi/2\phi_0)$ and inversion of (13) is no less difficult than cross-plotting, this latter method may be resorted to to obtain $K(W/d, W/S)$. In a certain range of values of W/d and W/S however, the work is lightened by a simplification. Our solution is strictly valid only when E^2 at a point directly under the middle of the bar has attained a substantial part of the value (ϕ_0^2/d^2) which it attains well under the infinite bar contemplated. In this range of ψ (13) and (15) have practically their asymptotic values:

$$K\left(\frac{\pi\phi}{2\phi_0}\right) = \frac{\dfrac{\pi\psi}{2\phi_0} - \log\sqrt{1+a}}{\dfrac{\pi\psi}{2\phi_0} + \sqrt{a}\,\mathrm{Tan}^{-1}\sqrt{a} - \log\sqrt{1+a}} \tag{18}$$

$$\frac{\pi x}{2d} = \frac{\pi\sqrt{a}}{2}\left(1+\frac{W}{S}\right) = \frac{\pi\psi}{2\phi_0} + \sqrt{a}\,\mathrm{Tan}^{-1}\sqrt{a} - \log\sqrt{1+a} \tag{19}$$

so that K may be directly expressed as follows:

$$K\left(\frac{W}{d}, \frac{W}{S}\right) = 1 - \frac{2}{\pi}\,\frac{1}{1+W/S}\,\mathrm{Tan}^{-1}\left(\frac{W}{2d}\cdot\frac{S}{W}\right). \tag{20}$$

As $d\to 0$ this gives $K=W/(W+S)$, which is proper.

NOTE 2

Derivation of Equation 4 for the Pressure Generated by the Thermophone

In addition to those already listed in Sect. 8, the following symbols are employed:

T = Temperature of the gas = $T_0 + T_1 e^{i\omega t}$

T_s = Temperature of the thermophone strip = $T_{s0} + T_{s1} e^{i\omega t}$

p = Pressure in the gas = $p_0 + p_1 e^{i\omega t}$

R = Gas constant.

Starting with the equation for the heat conduction in the gas:

$$k\frac{\partial^2 T_1}{\partial x^2} - i\omega C_p T_1 + i\omega \frac{C_p - C_v}{R} p_1 = 0, \tag{1}$$

we shall consider the one-dimensional problem in which the strips and plane boundaries of the enclosure are assumed to be of infinite extent. Let x represent the distance normal to the strip, and let d represent the distance between strip and diaphragm and between strip and mounting plug (metallic). The general solution of (1) is:

$$T_1 = A \text{ Exp } \alpha(1 + i)x + B \text{ Exp } [-\alpha(1 + i)x]. \tag{2}$$

In addition to the boundary condition

$$T_1 = T_{s_1} \qquad \text{when } x = 0$$

employed by Wente, we shall introduce the condition that at the metallic walls of the enclosure the variable part T_1 of the temperature is zero, that is:

$$T_1 = 0 \qquad \text{when } x = d$$

Evaluating the constants A and B we get

$$T_1 = \frac{(T_{s1} - mp_1)(e^{\lambda(x-d)} - e^{\lambda(d-x)}) + mp_1(e^{\lambda x} - e^{-\lambda x})}{e^{-\lambda d} - e^{\lambda d}} + mp_1 \tag{3}$$

where $m = (C_p - C_v)/R\rho C_p$.

Assume the pressure to be constant throughout the enclosure and the walls to be perfectly rigid, and also that the pressure may be calculated from the average temperature T_1 from

$$p_1/p_0 = \overline{T_1}/T_a \tag{4}$$

that is:

$$p_1 = \frac{p_0}{VT_a} \iiint T_1 \cdot dv. \tag{5}$$

In order to facilitate the solution of equation (1) we have assumed that the thermophone strip, parallel to the diaphragm and mounting plug, was of infinite extent. In an actual construction the strips are not only of infinite extent but may have substantially less area than the diaphragm, in which case the distribution of T_1 throughout the whole enclosure is no longer given by the above simple analysis. In view of the difficulty of exact calculation of these fringe effects an approximation must be resorted to, to be justified by experimental comparison with other absolute methods of calibration (e.g. Rayleigh disk, electric grille, pistonphone, etc.). As obvious approximation is to assume that the temperature in the region (1) directly between the foil and flat walls is that given by the one-dimensional solution (3) and that outside this region the temperature is zero. The average temperature over the whole enclosure will then be:

$$\bar{T}_1 = \frac{V_1}{Vd}\int_0^d T_1 dx = \frac{2S}{V}\int_0^{V/2S} T_1 dx \tag{6}$$

where S is the area of one side of the foil and V_1 is the volume of the region (1). This is the approximation employed, without discussion, by Arnold and Crandall (loc. cit.). It is evidently more accurate at high frequencies, when the thermal wave is confined to the neighborhood of the strip, than at low frequencies.

Adopting (6), performing the integration, solving for p_1 and then for T_{s_1} in terms of p_1 we have:

where

$$\begin{aligned} T_{s1} &= p_1 m\left\{2 - \frac{V\zeta\lambda}{2S}\left(1 - \frac{T_0}{P_0 m}\right)\right\} \\ \zeta &= (e^{-\lambda d} - e^{\lambda d})/(2 - e^{-\lambda d} - e^{\lambda d}) \\ \lambda &= \alpha(1 + i). \end{aligned} \tag{7}$$

The temperature of the strip T_{s_1} must now be calculated in order to express p_1 in terms of the known, or "impressed," quantities. Neglecting the loss of heat from the strip by radiation and conduction to the supports we have the following equation for the strip temperature:

$$-2kS\left(\frac{\partial T_1}{\partial x}\right)_0 + i\omega C T_{s1} = 0.48 I_0 I_1 R. \tag{8}$$

The derivative $\partial T_1/\partial x$ is to be evaluated by differentiating (3) and substituting $x=0$. Substituting (7) in (8) so expanded, and solving for p_1 we obtain

$$p_1 = \frac{.48 I_0 I_1 R}{m\left\{\left[2 - \frac{V\zeta\lambda}{2S}\left(1 - \frac{T_a}{mP_0}\right)\right][i\omega C - 2kS\lambda\theta] - 2kS\lambda/\zeta\right\}} \tag{9}$$

where

$$\theta = (e^{\lambda d} + e^{-\lambda d})/(e^{-\lambda d} - e^{\lambda d}). \tag{10}$$

Let us examine the quantities: θ and ζ. In a typical case for air $\alpha = 3.4f^{1/2}$ and for hydrogen $\alpha = 1.4f^{1/2}$. At 36 cycles and for $d = 0.5$ cm, $d\lambda = d\alpha(1+i) = 10(1+i)$. Since $e^{10} = 2.2 \times 10^4$, $\zeta = 1$, $\theta = -1$ with negligible error. Making these assumptions in (9) and rationalizing we obtain the expression (4) for the maximum value of the alternating pressure developed in the thermophone enclosure given in Sect. 8.

Note 3

Pressure Developed by the Pistonphone

Employing the symbols listed below equation (17), Sect. 16, let the piston be vibrated sinusoidally so that the displacement is $\zeta = \zeta_1 \sin wt$.

The diffusion of heat in the gas is subject to the equation:

$$k\nabla^2 T_1 - i\omega C_p T_1 + i\omega p_1 \frac{C_p - C_v}{R} = 0. \tag{1}$$

The walls will be far apart compared with the "wavelength" of the diffusive thermal wave so that if x represents the distance from the walls:

$$T_1 = mp_1[1 - e^{-\alpha x(1+i)}] \tag{2}$$

since $T_1 = 0$ when $x = 0$ and $T_1 = mp_1$ when x is large. Here

$$\alpha = (\omega\rho C_p/2k)^{1/2}; \qquad m = (\gamma - 1)T_0/\gamma P_0.$$

For a perfect gas:

$$dp/p_0 = dT/T + dV/V \tag{3}$$

or for the alternating components:

$$p_1 = P_0 \cdot \overline{T}_1/T_0 + \zeta_1 S_p P_0/V. \tag{4}$$

$\overline{T}_1$ represents the average value of the temperature amplitude. The average temperature up to a distance d from the walls would be:

$$\overline{T}_1 = \frac{1}{d}\int_0^d T_1 dx = mp_1\left[1 - \frac{1}{d\alpha(1+i)}\right]. \tag{5}$$

What value shall be assigned to d? Consider a narrow section with parallel walls of total area S_w separated by $2d$. Then the volume will equal $2S_w d/2$ and $d = V/S$. Extending this idea to an enclosure of large volume and of simple shape we obtain:

$$\begin{aligned} p_1 &= \frac{p_1(\gamma - 1)}{\gamma}\left[1 - \frac{S_w}{\alpha V(1 + i)}\right] + \frac{P_0 S_p \zeta_1}{V} \\ &= \frac{\gamma p_0 S_p \zeta_1}{V_1}\left\{\frac{1}{1 + \dfrac{(\gamma - 1)S_w}{\alpha V} + \dfrac{(\gamma - 1)^2 S_w^2}{2\alpha^2 V^2}}\right\}^{1/2}. \end{aligned} \tag{6}$$

APPLICATION OF NORRIS-ANDREE METHOD OF REVERBERATION MEASUREMENT TO MEASUREMENTS OF SOUND ABSORPTION

By R. F. Norris

C. F. Burgess Laboratories, Inc., Madison, Wisconsin

In acoustical work it is often inconvenient and costly to make large samples of experimental materials, although it may be imperative to know their sound absorbing coefficients. Most of the methods of sound absorption measurement in use require relatively large samples and

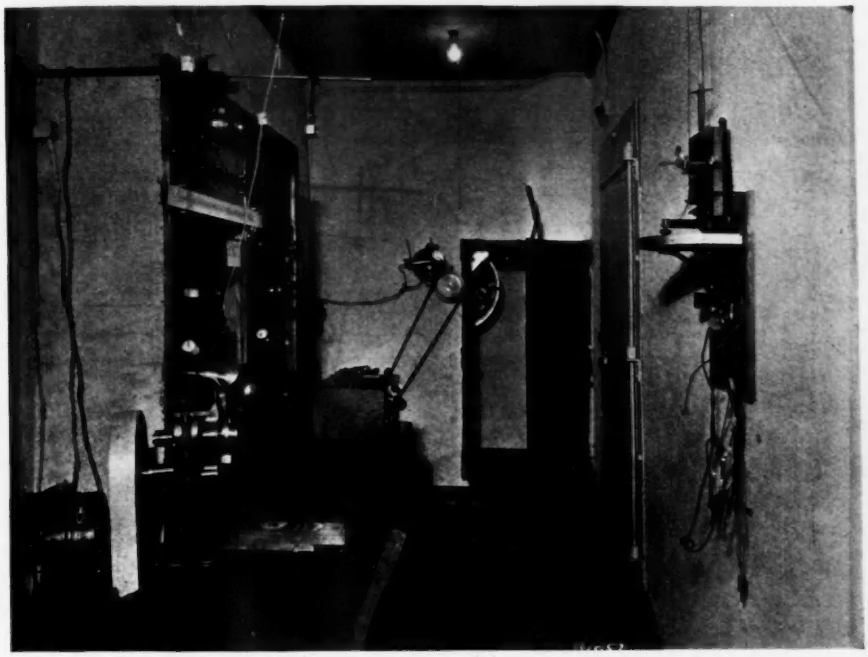


Plate I. *In this general view of the control room the rotary switch is shown in the foreground. At the rear center the door to the test chamber is open showing the microphone support. The white spot is the edge of the mixing paddle.*

consequently a new method was sought which, while being adapted to the use of small samples, would yield accurate and reproduceable results.

Since the Norris-Andree method of reverberation time measurement will measure extremely short periods of reverberation, it should give

accurate results in small rooms. For this reason it was chosen as the basis of the absorption measurement scheme developed at the Burgess Laboratories.

A chamber was laid out five feet by six feet by nine feet high. The walls are of eight inch concrete finished in an extremely hard surfaced gypsum plaster. It is provided with a paddle mounted longitudinally,

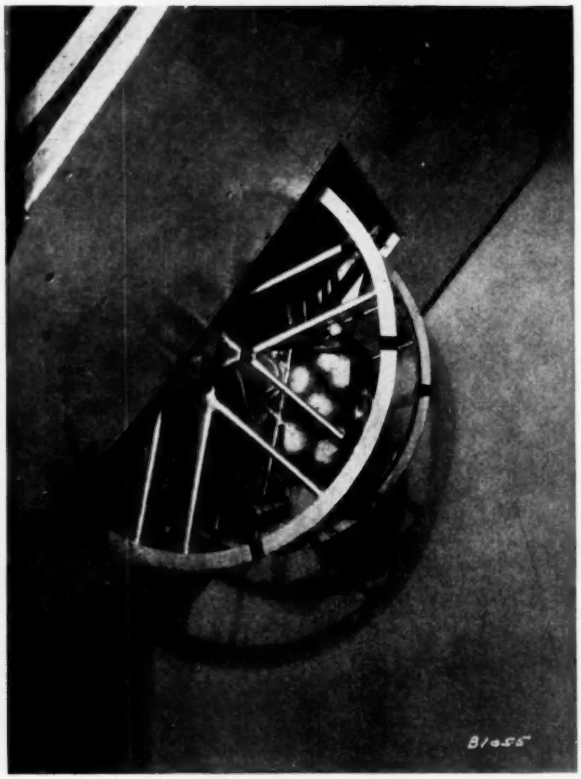


PLATE II. *The microphone and speaker support is built like a spoked wheel for convenience. The method of slotting the paddle to allow these units to be mounted at the center of the chamber is clearly shown.*

with the shaft parallel to the five foot dimension and mounted in the center of the six foot sides. The paddle is four feet six inches wide and five feet six inches long and has an opening in the center in which a microphone and dynamic speaker are mounted in a supporting framework carried by a stationary hollow shaft which enters through one end of the paddle shaft. This hollow shaft also acts as a conduit for the wiring for both the speaker and the microphone. (See plate II.) The paddle

is arranged to be driven by a motor with speed control, located outside of the test chamber. The speed of the paddle is indicated by a stroboscopic wheel so placed that the operator can check it conveniently. See plate III.

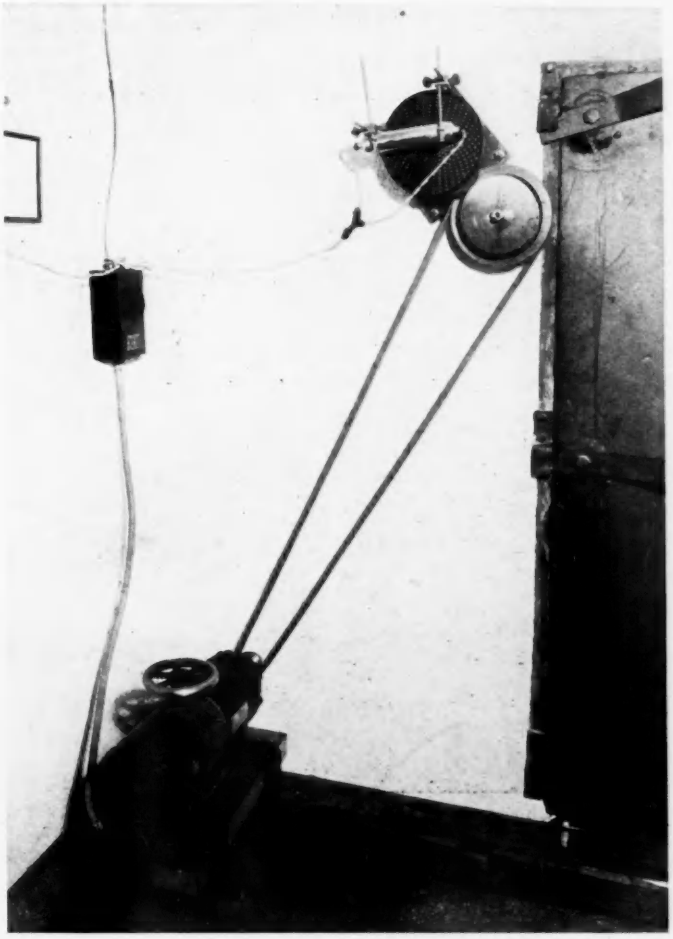


PLATE III. *The mixing paddle is driven by a unit motor and speed changer through a V belt. The spotted disk, when illuminated by a neon glow lamp operated by a 60 cycle current, acts as a stroboscope and indicates the speed of the paddle.*

The system supplying oscillating current to the speaker consists of an oscillator so keyed by a rotary cam switch that it operates over 15 degrees of the full revolution and is inoperative during the remaining 345 degrees. See plate IV. The output of this oscillator is fed to the speaker

through a filter. This causes the speaker to sound the note to which the oscillator and filter are tuned at definite intervals and during small uniform periods of time.

The measuring system comprises a microphone, a filter, a rotary cam switch and amplifier and a ballistic meter. The microphone picks up the sound continuously and feeds it to the filter. The rotary cam switch is connected between the filter and the amplifier and is so arranged that the current from the filter is allowed to enter the amplifier only during

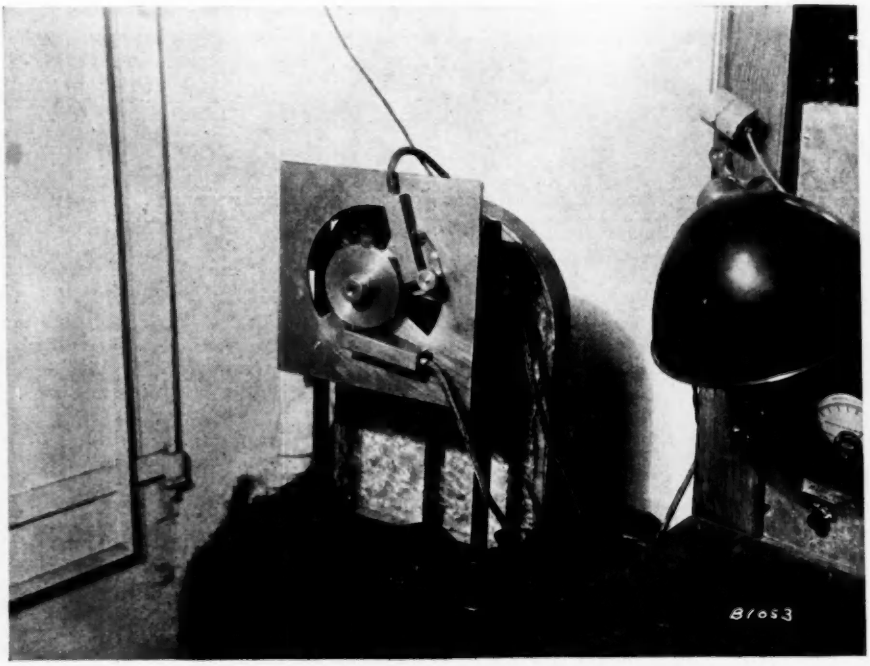


PLATE IV. *The cam switch consists of two cams, both mounted on the same shaft but operating different switch jacks. The lower jack, which is stationary, operates the oscillator, while the upper may be rotated and controls the amplifier.*

definite time intervals. By rotating this switch around the cam these intervals may be made to start directly after the speaker stops or at any length of time later up to 3.4 seconds.

Since the switch cam rotates once in 3.6 seconds, each degree on the calibrated switch support represents 1/100 of a second and the time lag between the cut off of the source and the beginning of the measuring period may be read directly on the scale.

The two cams, one controlling the speaker and the other controlling

the amplifier, are mounted on the same shaft which is driven at a constant speed by a synchronous motor acting through a gear train.

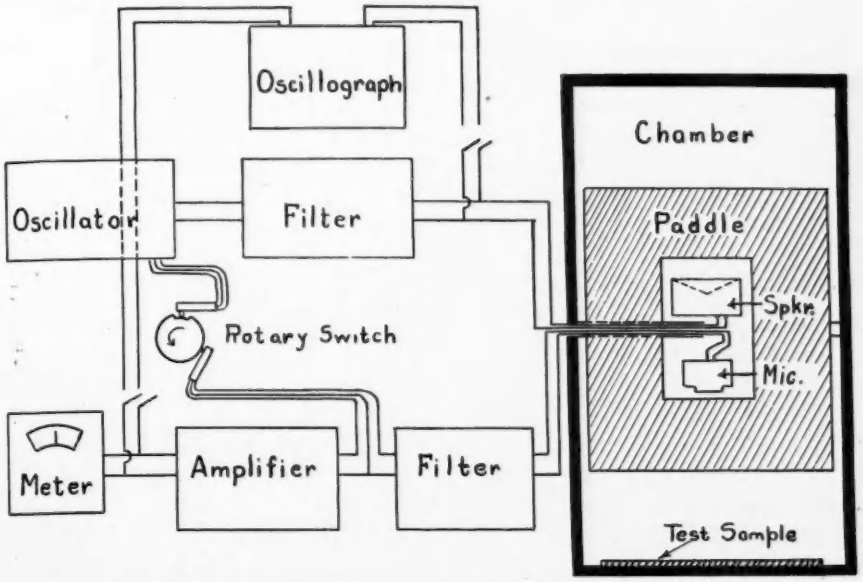
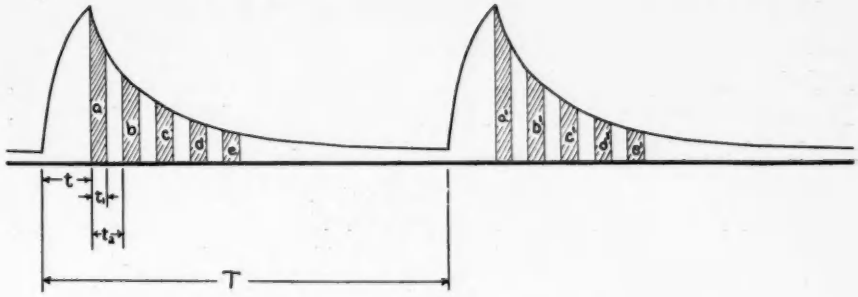


PLATE V. *This diagram shows schematically the electrical set up of the equipment.*

To check on the wave form entering the chamber and to compare it with that picked up by the microphone a two element oscillograph is



T = One revolution of the switch cam
t = Time source speaks
t_1 = Interval over which measurement is made
t_2 = Time shift between successive measurements

PLATE VI

arranged so that it may be connected into the two circuits simultaneously. The waves may thus be studied, compared and if any irregulari-

ties of form appear adjustments may be made to correct them. See plate V.

The cycle of operations may best be shown graphically. See plate VI. The total time of one revolution of the switch cam may be represented by T which in this apparatus is 3.6 seconds. Thus the time during which the source is energized is t or in this instance .15 seconds. The length of time the amplifier is in use is t_1 which is controlled by the length of the depression in the cam in this case .05 seconds. The time interval between the stopping of the source and the beginning of the measurement period may be adjusted to any value from 0 to 3.4 seconds. In making measurements it is usually started at 0 and increased in .05 second intervals up to .3 seconds.

METHOD OF MAKING READINGS

The apparatus is started and allowed to warm up for 20 minutes. The oscillator is set to produce the frequency at which measurements are to be made and the two filters are set to this same frequency. The input and output frequencies are studied while the paddle is stationary and the cam switch is shorted out keeping both the oscillator and amplifier on steadily. To obtain a calibration reading the amplifier jack on the cam switch is shorted while the oscillator jack operates normally. The meter then assumes a constant reading which is noted down every 15 seconds for 10 readings. The amplifier cam switch is thrown into action and the cam is set at 0° on the graduated arc. At each sounding of the source a reading is observed on the meter. This reading is recorded each revolution of the cam until 10 readings have been taken. The amplifier cam switch is set back 5° and a second series of 10 readings is taken. This set of readings will be lower than the first since the measurement area is taken later on the decay curve. The cam switch is then set to 10° and the operation repeated. When readings have been taken over 30° on the cam switch the amplifier is again set to operate continuously and a second calibration reading is taken. If nothing has changed this reading will check the first reading exactly. If there has been a change and it is small the averages of the consecutive readings may be each corrected proportionally.

The entire proceeding is then repeated after the sample to be tested is placed on the floor of the chamber, and a second set of readings is obtained.

The data as taken and averaged may be plotted directly on semi-log paper. The points should fall on a straight line. If they do not, readings

should be discontinued and the apparatus checked over to find the part that is not operating properly.

To determine the absorption of the material tested the time of reverberation of the chamber, empty and with the sample, is computed by extending the curve and measuring the time taken for a 60 db. drop. These times are substituted in the following equation which may then be solved for alpha:

$$\log_{10}(1-\alpha_s) = \frac{24V}{Av}\left(\frac{1}{t_\beta} - \frac{1}{t_\alpha}\right) - \frac{24V}{Sv}\left(\frac{1}{t_\beta}\right).$$

Where α_s = the absorption coefficient of the material,

V = the volume of the chamber.

A = the area of the sample.

v = the velocity of sound.

S = the total surface of the chamber.

t_β = the reverberation time of the empty chamber.

t_α = the reverberation time of the chamber with the sample in place.

This equation is derived as follows:

Let I_0 be the original intensity of the sound,

I_n be the intensity after n reflections,

Then

$$I_n = I_0(1-\alpha)^n. \tag{1}$$

If t is the time required for n reflections then,

$$n = \frac{t}{\frac{4V}{S}} \cdot \frac{1}{v} = \frac{Svt}{4V}, \tag{2}$$

and equation (1) may be written:

$$I_n = I_0(1-\alpha)^{\frac{Svt}{4V}}. \tag{3}$$

In calling t the period of reverberation we assume a 60 db. drop between I_0 and I_n and

$$\frac{I_n}{I_0} = (1-\alpha)^{\frac{Svt}{4V}} \tag{4}$$

becomes

$$-6 = \frac{Svt}{4V}\log_{10}(1-\alpha), \tag{5}$$

or

$$\log_{10}(1-\alpha) = -\frac{24V}{Svt}. \tag{6}$$

If we let β = the average absorption coefficient of the empty chamber and

α_s = the average absorption coefficient of the sample of area A, we may write

$$S \log_{10}(1-\beta) = -\frac{24V}{v}\cdot\frac{1}{t_\beta} \tag{7}$$

for the absorption of the empty chamber and

$$A \log_{10}(1-\alpha_s) + S \log_{10}(1-\beta) - A \log_{10}(1-\beta) = -\frac{24V}{v}\cdot\frac{1}{t_\alpha} \tag{8}$$

for the absorption of the chamber with the sample.

Subtracting (7) from (8) we arrive at

$$A \log_{10}(1-\alpha_s) - A \log_{10}(1-\beta) = \frac{24V}{v}\left(\frac{1}{t_\beta} - \frac{1}{t_\alpha}\right) \tag{9*}$$

for the absorption coefficient of the sample which may be written

$$\log_{10}(1-\alpha) = \frac{24V}{Av}\left(\frac{1}{t_\beta} \quad \frac{1}{t_\alpha}\right) \quad \frac{24V}{Sv}\cdot\frac{1}{t_\beta}. \tag{9}$$

The accompanying curves are from a series of tests run on an acoustical plaster. See plate VII. The values shown as β are the average absorption coefficient of the chamber surfaces at the various frequencies while α is the average absorption coefficient of the samples tested. The value shown as S is the value of absorption of the sample computed according to Sabine's formula and is given because of the wide use of this formula in the literature.

In using this method of test certain precautions must be observed.

For instance it was found that if the rotary switch was made to break the circuit between the oscillator and the speaker, a loud thump was produced just as the sound was cut off. This thump interferred with the readings especially at the higher frequencies. It was eliminated by shutting off the oscillator by short circuiting the feed back circuit and not breaking the circuit between it and the speaker.

* NOTE: $2.3\times-\log_{10}(1-\alpha_s)$ = Sabine coefficient (a). Therefore, if the right hand part of this equation is evaluated, its sign changed and multiplied by this factor, (a) may be found directly.

It was found that the paddle was necessary if readings that meant anything at all were to be taken. The accompanying curves show what resulted it the paddle was stopped either in the horizontal or vertical positions. See plate VIII. A similar result is obtained if the ratio of the speed of the paddle to that of the time switch is unity or a small whole number. In the chamber under discussion the cam switch runs at 16–2/3 RPM while the paddle runs at 46.15 RPM.

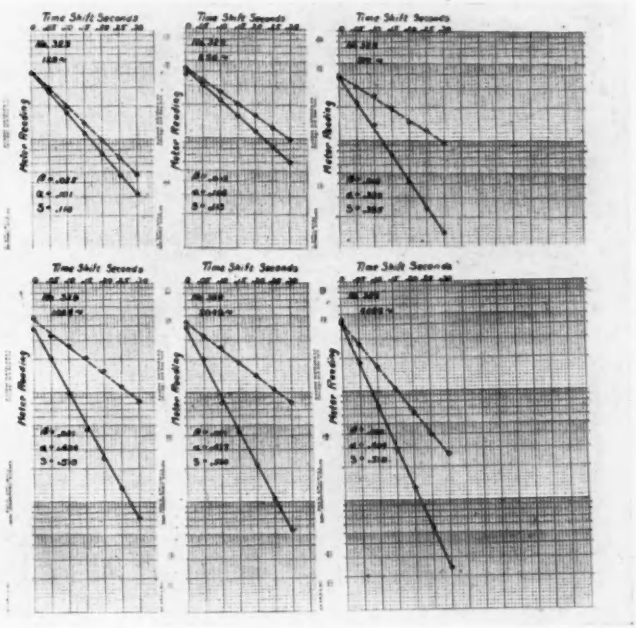


PLATE VII

At certain frequencies, depending on the dimensions of the chamber, resonance may occur. In this chamber such an effect was found at 128 and 256 cycles. The accompanying series of curves show that while at 250 cycles good readings may be made, reliable results are not to be obtained at 256 cycles while conditions are worse at 258 cycles. See plate VIII.

At 270 cycles (not shown) the points again fall on a straight line. For this reason data is taken at 120 and 250 cycles instead of 128 and 256 cycles.

Standard materials have been remeasured form time to time and the various values obtained have not differed by more than ± 3 per cent. Some of this variation is undoubtedly due to the method of measurement while a good part of it is probably due to changes in the samples.

The absorption of the empty chamber apparently changes slightly from time to time but always averages about the same. Some work has been done to find the cause of these changes but so far no conclusive results have been obtained. Apparently, neither temperature nor humidity changes are the cause and at present an attempt is being made to correlate the changes with barometric pressures.

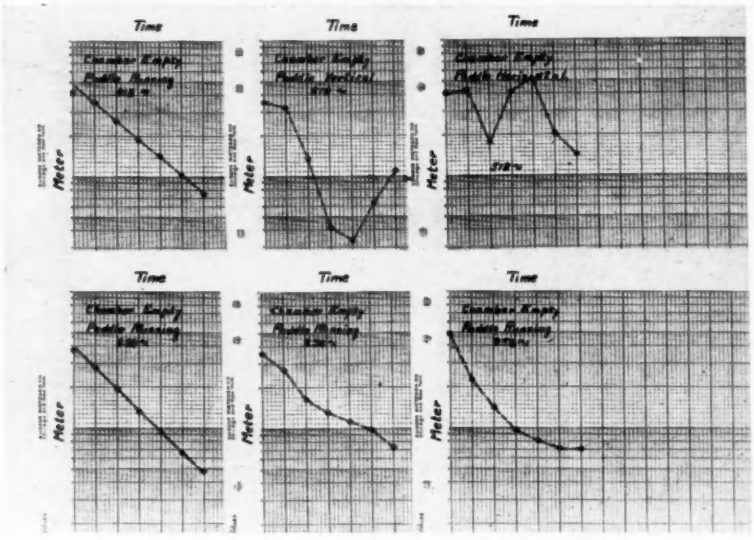


PLATE VIII

The uniformity of results with this method of measurement is particularly gratifying since the chamber is located in the basement under a machine shop and no attempt has been made to isolate it from the rest of the building.

Tests made while the shop is in operation show no difference in results from those made at night when the building is absolutely quiet.

The advantages of this method of test are obvious. They are:

1. Only a small sample of the material to be tested is needed. Sixteen square feet being used in the chamber under discussion.
2. Check results are easily and quickly obtained.
3. The chamber may be small and need not be isolated from the rest of the building.
4. No reference material is needed against which the unknown may be calibrated.
6. The data by its deviation from a straight line when plotted warns the operator if the apparatus is not running properly.
7. The computation of the absorption coefficients is simply and quickly done.

THE NOISE SURVEY OF THE RAPID TRANSIT LINES OF NEW YORK CITY*

By G. T. Stanton and J. E. Tweeddale
Electrical Research Products, Inc.

Rapid transit lines are generally admitted to be serious sources of disturbing noise. The general noise survey for the Noise Abatement Commission of the City of New York showed the subway and elevated lines of that city to be major contributors, if not the most serious addition to the city's din. The importance of noise as a factor in the possible future depopulation of cities, and recognition of the huge economic losses which it causes, has created strong public sentiment for noise abatement programs.

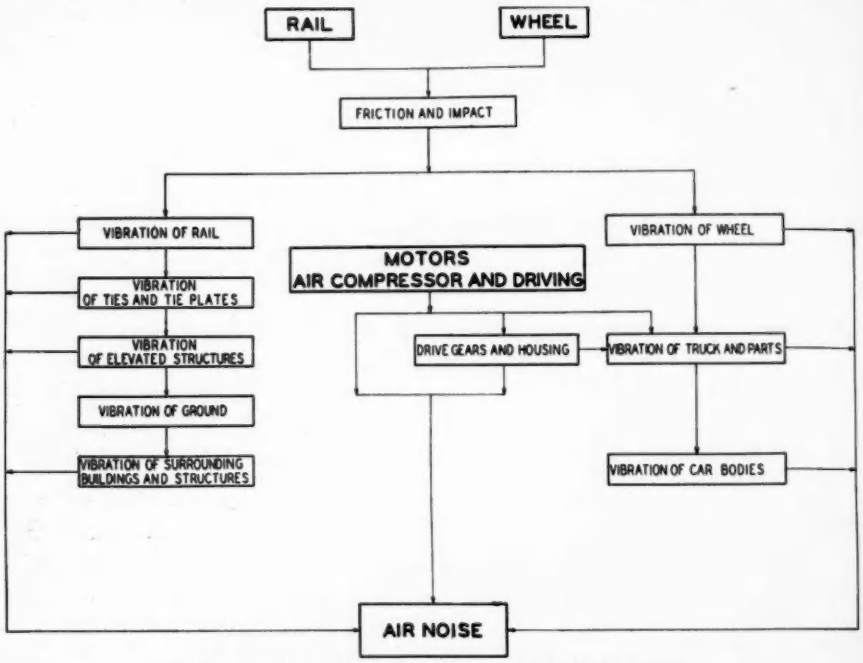
The reduction of noise in any rapid transit line is a major problem, involving many factors both economical and mechanical. When it is appreciated that the investment in such facilities in New York City approximates the enormous sum of one thousand million dollars, the need for a carefully planned program, based upon the most critical of studies, is evident. The Board of Transportation, therefore, inaugurated a survey of the actual noise conditions in existing lines, as a preliminary to the formulation of a policy and program for the ultimate solution of this problem.

The creation of excessive noise and its effect on passengers are objections common to both subway and elevated lines. The depreciation in value of surrounding property is, however, almost exclusively a problem of the elevated. The abandonment of existing elevated lines and the construction of all future lines in subways would involve tremendous loss in invested capital and require expenditures several times greater than for elevated structures in outlying undeveloped areas. The fundamental purpose of the survey was to determine from existing conditions, which of the existing types of elevated structures created the least disturbance, whether or not such structures would be suitable for future extensions, and the feasibility of obtaining sufficient reduction in noise level in areas adjacent to elevated structures to warrant their serious consideration in future extensions. A secondary purpose was to determine the possibility of immediately reducing noise conditions a perceptible amount by means short of drastic changes or large expenditures. To be able to satisfactorily answer these questions, it was neces-

* For presentation at meeting of the Acoustical Society of America-Cleveland, November 30–December 1, 1931.

sary to determine in a definite quantitative manner, the amount of disturbing noise produced by the elevated operation, the relative loudness of noise created by different types of structures and cars, and the possibility of developing improved types of construction and equipment which will produce so little noise as not to be objectionable. The general results and conclusions obtained thus far are presented in this paper.

A large number of possible combinations of noise sources and influencing factors contribute to the overall noise created by rapid transit opera-



GENERAL ANALYTICAL DIAGRAM SHOWING THE PATH OF TRANSMISSION OF VIBRATION ENERGY AND THE RADIATION OF NOISE FROM AN ELEVATED TRAIN AND STRUCTURE

FIG. 1.

tion. A careful preliminary analysis of all noise sources was required to determine the procedure for the collection of the data, in order to acquire the information desired from the field survey. Observation indicated that the primary noise source caused by the passage of trains over rail structures was due to the impact and friction of car wheels and flanges on the rail itself. The impact and friction created both noise and vibration energy. Analysis inferred that this vibration energy divided into two general paths from this point of contact before being radiated as audible noise. These two general paths were the elevated structure

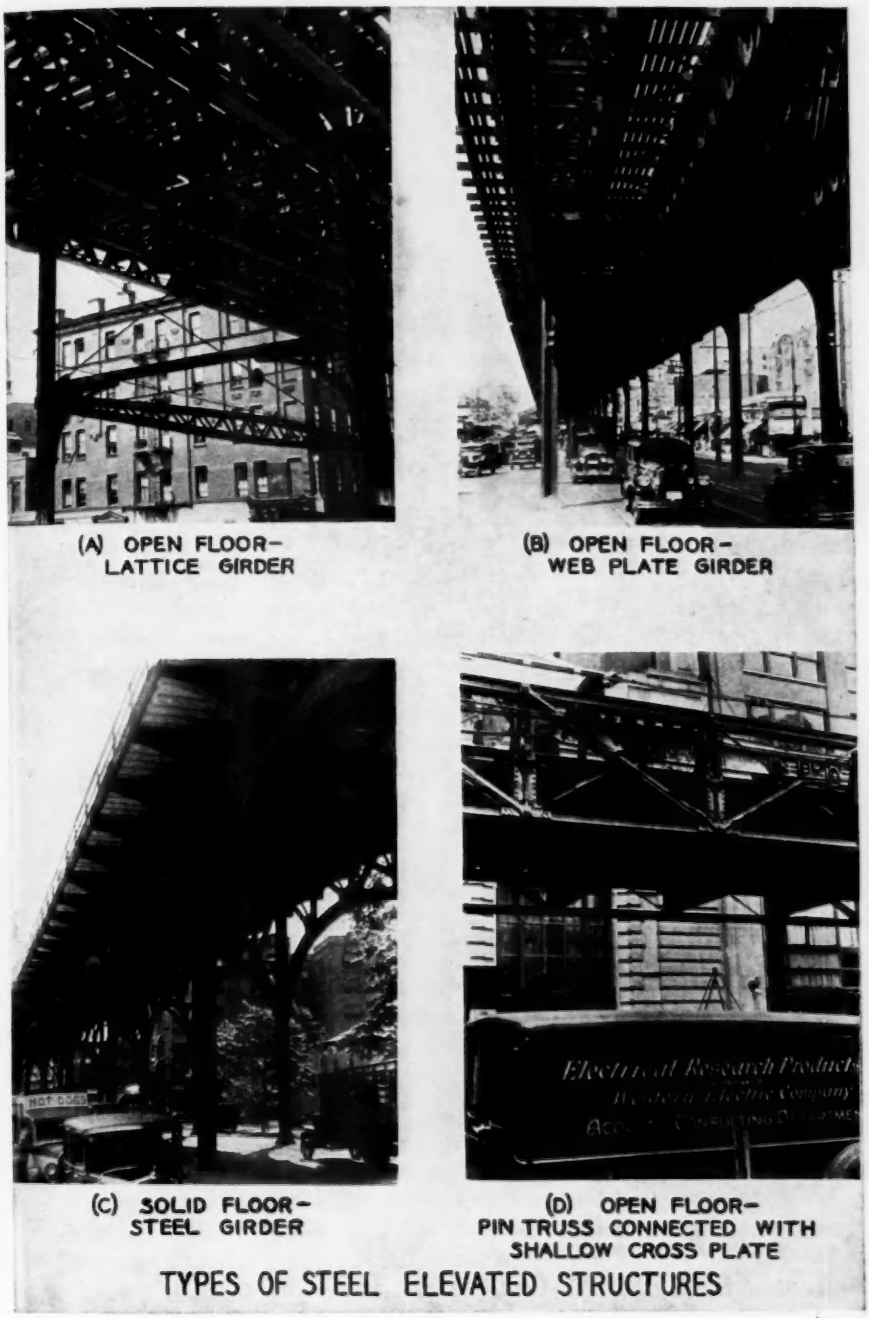

(A) OPEN FLOOR–
LATTICE GIRDER
(B) OPEN FLOOR–
WEB PLATE GIRDER
(C) SOLID FLOOR–
STEEL GIRDER
(D) OPEN FLOOR–
PIN TRUSS CONNECTED WITH
SHALLOW CROSS PLATE
TYPES OF STEEL ELEVATED STRUCTURES

FIG. 2.

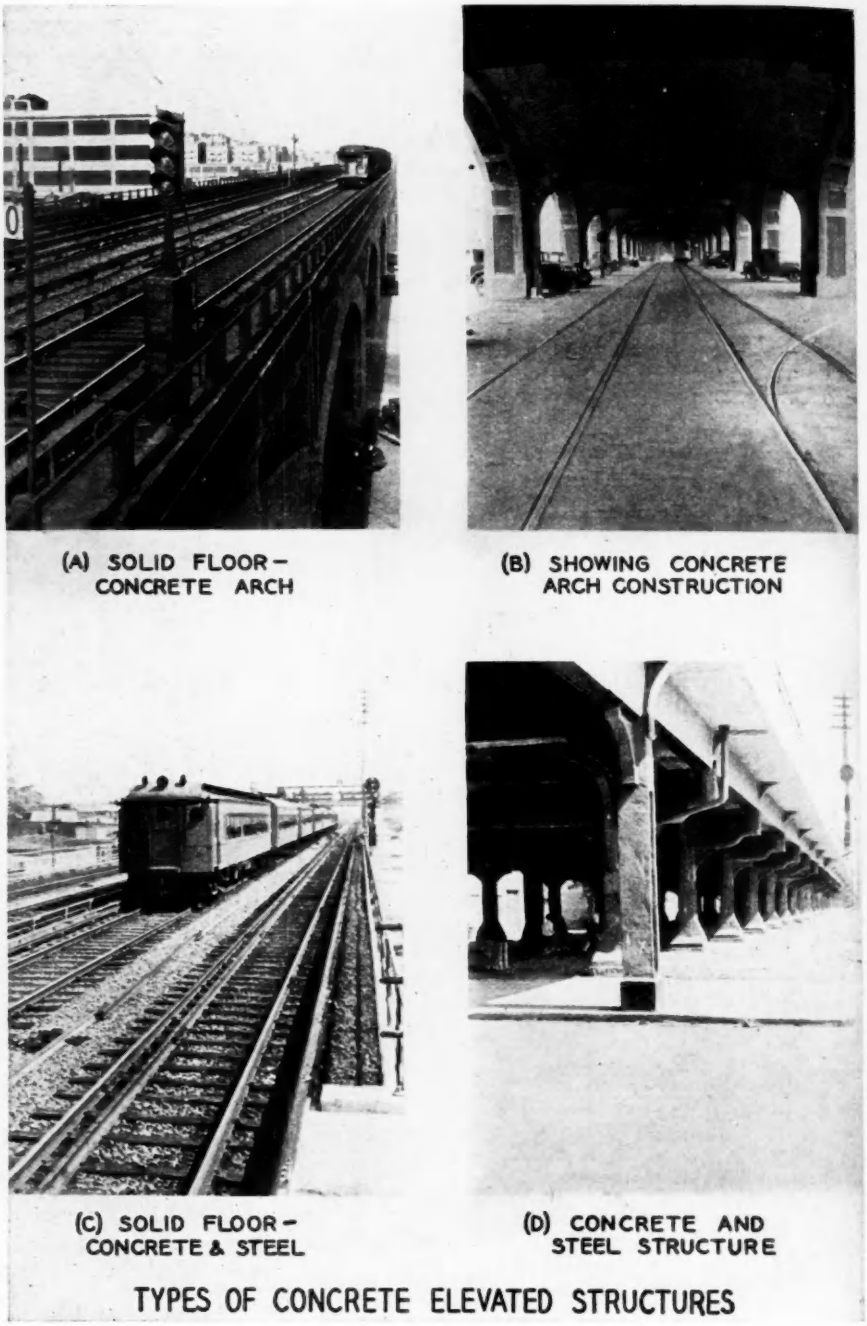

(A) SOLID FLOOR – CONCRETE ARCH
(B) SHOWING CONCRETE ARCH CONSTRUCTION
(C) SOLID FLOOR – CONCRETE & STEEL
(D) CONCRETE AND STEEL STRUCTURE
TYPES OF CONCRETE ELEVATED STRUCTURES

FIG. 3.

and car body, as are indicated in Figure 1. It is apparent, therefore, that variations in the nature of these two paths and their component parts would directly influence the amount of noise finally radiated. In addition, there were various primary noise sources inherent within the motive power equipment of the cars themselves.

Examination of the structures revealed that there are four general types in common use: pin-truss connected with shallow cross plate, lattice girder, web plate girder, concrete arch and concrete and steel. The web plate girder constructions had both open and closed track floors, and in the concrete arch and the concrete and steel structures, the floor was always of concrete with a rock ballast roadbed. In Figures 2 and 3 are shown typical examples of these general types of structures. Several types of steel and wooden cars are in use on these structures, all of sufficiently different construction to warrant individual investigation of the noise created by their operation. Also, the motive power equipment is of great variety with both open and cased spur gear drives. In addition, maintenance of car, structures, track, etc., was anticipated to have some influence on the creation of noise. It was recognized that operating conditions and schedules; that is, make-up of trains, the number of motor and trailer cars, operating speeds, rates of acceleration and deceleration would be directly involved in the amount of noise created for any particular combination of circumstances. In addition, the nature of surroundings would influence the results.

It was apparent, therefore, that observational data would have to be obtained in a regular and constructive method to permit the breakdown and evaluation of the importance of each of the variable factors. In addition, the data would have to be obtained in such a manner as to permit the determination of the probable noise conditions under combinations of factors that did not exist on any of the present lines. Consequently, the test procedure was prepared on the basis of holding constant as many factors as possible, allowing but one to vary at a time. For example, measurements were made of the noise created by different types of trains of constant composition operating under equivalent conditions, over essentially similar track construction and maintenance on similar elevated structures in equivalent surroundings. Following this, the procedure of the survey was to make observations of the effect of variation of types of structures using types of trains previously observed, all other factors being held constant. This part of the survey was divided into a comparison of the effect of open and closed track floors. In all, it was anticipated that forty to fifty locations would be

necessary to establish representative conditions that would permit complete analysis of the influencing factors with a sufficient number of check observations to determine the limits of accuracy of the observations. It was found that since measurements must be made under operating conditions without special test trains in the majority of cases, that reasonable assumptions of acceleration and speed would have to be made from the operating schedules. Subsequent measurements of these factors in a number of cases substantiated the validity of the various assumptions made.

It was anticipated prior to the start of these observations that the location of the microphones used for picking up the noise would affect the interpretation of the data obtained. It was assumed that if a microphone were placed on the structure close to the passing trains, that the noise level measured would be primarily caused by the cars and their inherent influencing factors. At the same time, measurements of noise level made simultaneously at street level under the structure and at the building line would be more largely influenced by the structure and its component factors. Measurements indicated the correctness of these assumptions in the majority of cases, and allowed a direct indication of the trend in noise level caused by either of the two radiating sources. Wherever peculiar formation of structure or surroundings indicated possible difficulty in the segregation of the contributory factors, other microphone positions were taken in addition. It was anticipated that while the program for the survey of the various noise sources could be quite completely planned at the start, the variety of conditions would cause may modifications and extensive readjustments as the work progressed to satisfactorily permit the analysis of a number of unanticipated conditions. However, in spite of this difficulty, the preliminary planning did furnish a reliable basis for the determination of the more regular factors.

Measurements were made with two types of direct indicating instruments, a visual indicating meter and a recording meter. Both of these devices were used in conjunction with band pass filters to obtain a physical analysis of the general frequency composition of the various noise sources. The visual indicating meter consisted of a calibrated dynamic microphone, an amplifier with a frequency weighting network, a distortionless attenuator and a rectifying device with a meter. In these measurements a frequency weighting network based on the equal loudness curve at thirty sensation units was used. In addition, the damping factor of the indicating meter was adjusted to simulate the ballistic

characteristics of the average ear. The sound meter was calibrated to have a zero reference level corresponding to the threshold of audibility of average street noise. The complete description of this general type of

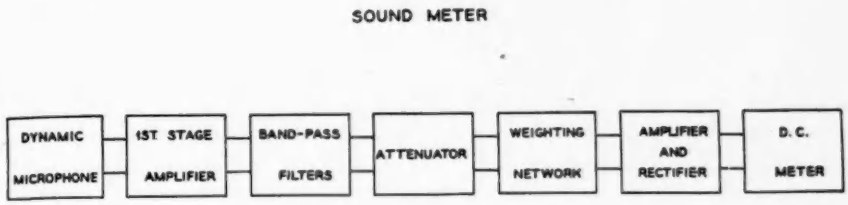


FIG. 4.

sound meter has been given in the recent literature.[1,2] A simplified schematic diagram of this meter used in conjunction with band pass filter banks is shown in Figure 4.

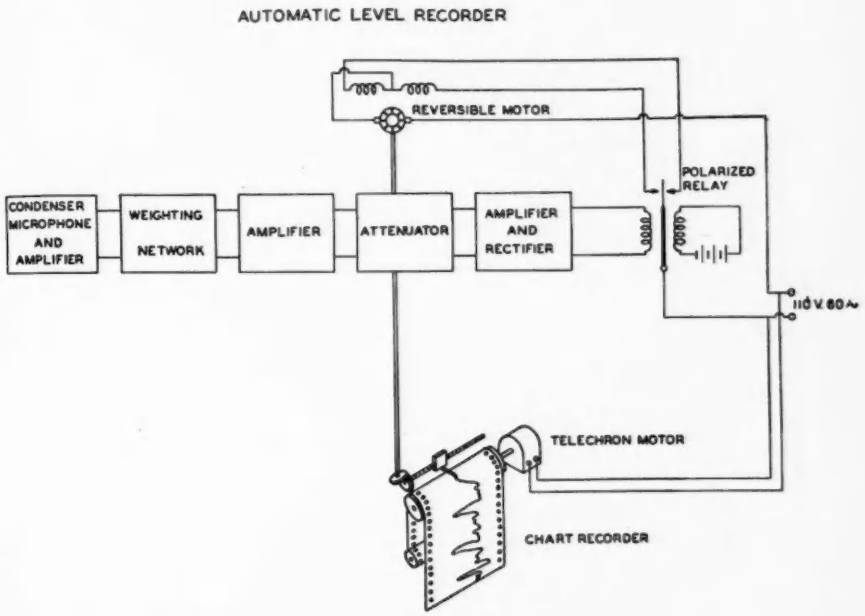


FIG. 5.

This type of sound meter was used most effectively for the rapid observation of a large number of noise sources of widely varying character and intensity. In measurements involving but one type of noise source

[1] Journ. Acous. Soc. Am. Vol 11. July 1930, p. 31.
[2] Electrical Engineering Vol. 50. No. 5, May 1931, p. 342.

and requiring no other observational data than the noise level, a recording device was used satisfactorily. This recording instrument consisted of a condenser microphone, attenuator, amplifier and rectifier with a polarized relay controlling a motor actuating the attenuator. The movement of the attenuator was recorded by a pen on a continuous chart driven by a telechron motor. Any change of noise level caused a variation in the rectified current supplied to the polarized relay. The variation in current about the balance point of the relay caused it to operate

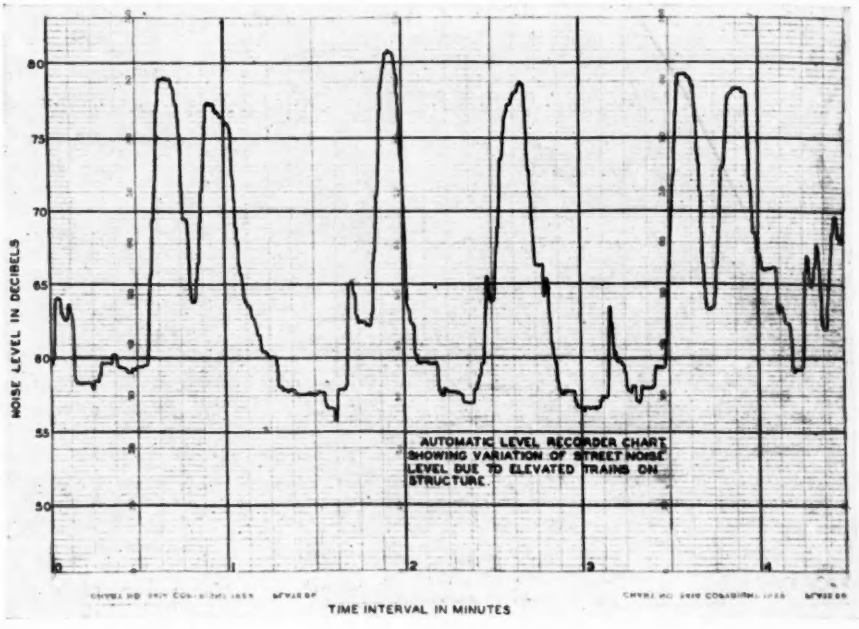


FIG. 6.

a reversible motor driving the attenuator control of the amplifier. The action of the device was to maintain a constant output to the polarized relay by varying the amplifier gain to correspond with the variation in noise level. For this survey the instrument was arranged to utilize the frequency weighting network of the sound meter previously described. A simplified schematic diagram of this instrument which is termed the "Automatic Level Recorder" is shown in Figure 5. A chart taken with the Automatic Level Recorder of the variation in street noise level due to passing elevated trains is shown in Figure 6.

As might be anticipated, the noise levels created by individual trains at any particular location had an appreciable deviation from the aver-

age results computed for the location. It was found though that at other locations having similar construction, surroundings and cars, and correcting for the variations in operating schedules, that the average results for each location checked with unusual consistency. Consequently, it was concluded that averages of large numbers of observations corrected to similar operating conditions would be completely representative of the prevailing noise conditions.

The summation of the results of the survey of the noise conditions of the elevated lines revealed that they created relatively higher noise levels compared with that of usual heavy street traffic. It was found that for average conditions, the noise level of trains operating on the elevated structure varied on the average from 83db. to 91db., this, in general, depending on the type of structure and type of car. These levels may be compared with the general street noise level in the immediate vicinity of the structure, varying from 60 to 70 db. The seriousness of the noise conditions produced in office and apartment buildings immediately adjacent to the elevated structure may be appreciated by the fact that measurements in these locations showed an average noise level of 75db. with all windows closed, and with windows open this noise level frequently reached peaks as high as 85db.

The study of the influence of the type of structure on the amount of noise created, revealed many interesting comparisons. It was found, in general, that less noise was created by the operation of trains under normal conditions on lattice girder structures than on any other type. Examination indicated that the operating speeds on lines using this type of structure were in general considerably less than on other types, and that for equal operating speeds and conditions this type of structure would be more noisy than the more rigid types. It was also found that substantially the same amount of noise was produced by trains operating on web plate girder structures using closed floor construction as that created by their operation on concrete arch construction. However, on the concrete arch construction it was found that the noise did not spread over as large a surrounding area. In both cases where the closed floor was used, i.e., web plate girder with solid floor and concrete arch with closed floor, considerably less noise was found to exist in the street under the structure. These findings refer to sections where no high buildings were located close to the elevated lines. In more congested areas with high buildings close to the elevated lines, the noise being confined between the buildings, was more generally distributed, and it was found that the shielding effect of closed floor structures was of less

value. The noise at the street level under these conditions was found to be substantially the same as if an open floor construction had been used.

TABLE I

AVERAGE NOISE LEVELS OF TYPE "A" CARS OPERATING ON ELEVATED STRUCTURES IN OPEN SURROUNDINGS AT A SPEED OF 20 M.P.H.

Structure	Track Floor	Average Overall Noise Level	
		On Structure	At Street Level
Lattice Girder	Open	85db.	84 db.
Web Plate	Open	91	87
Web Plate	Closed	89	79
Concrete Arch	Closed-Rock Ballast Bed	89	74
Concrete and Steel	Closed-Rock Ballast Bed	87	—*

* Less than traffic noise level of 65 db. on street.

In Table I is given a summary of the noise conditions created by the operation of one type of steel car at a speed of 20 m.p.h. over different types of elevated structures. These results in all cases refer to locations having substantially open surroundings.

Measurements of the different types of cars now in operation on elevated and subway lines showed appreciable differences in the loudness of noise created. In general, the somewhat heavier types of subway cars created a greater amount of noise than the lighter types of cars. As was anticipated, it was found that the number of cars per train had relatively small effect on the maximum loudness of the noise. It was found that there was quite a direct relationship between the speed of the trains and the noise they created. The construction of the cars composing these different test trains are as follows:

Type "A"—Old type, single shell, steel car
Type "B"—Modern, insulated double shell, steel car
Type "C"—New type, insulated double shell, steel car
Type "D"—Insulated double shell, articulated steel car sections (four trucks per three car section)
Type "E"—Old type, open platform, wooden elevated cars.

In Figure 7 are shown curves of noise level versus operating speed for Type "E" cars when operated on lattice girder structures. It will be seen that in general for every 5 m.p.h. increase of operating speed, approximately a 5db. increase in noise level occurred. It will also be observed that when coasting without power the noise level created was approximately

4db. less than when operated under power. This difference was undoubtedly due to the changes in the noise and vibration energy emitted by the motor and its influencing factors.

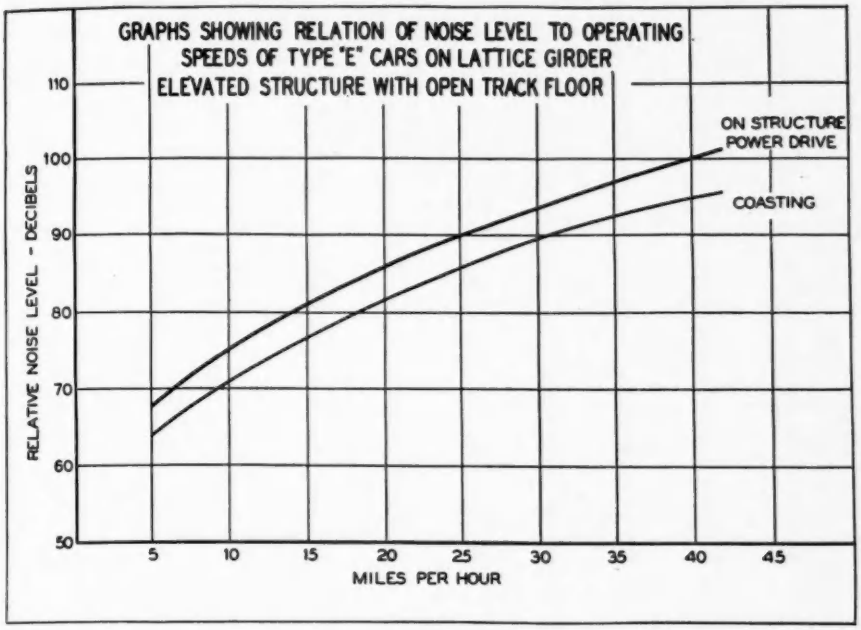


FIG. 7.

TABLE II

AVERAGE NOISE LEVEL OF VARIOUS TYPES OF STEEL CARS, ROCK BALLASTED ROADBED, NO STRUCTURE, OPEN SURROUNDINGS, AND OPERATING SPEED 40 M.P.H.

Car	Location	Average Noise Level
Type "A"	Exterior	90.5 db.
Type "B"	Interior: Windows Open	82.5
	Windows Closed	77.5
	Exterior	87.0
Type "C"	Interior: Windows Open	86.5
	Windows Closed	82.5
	Exterior	84.0
Type "D"	Interior: Windows Open	80.0
	Windows Closed	75.0
	Exterior	86.5

* Observations exterior to cars taken at edge of right-of-way—25 feet from center line of track.

In Table II is shown a summary of noise levels created by the operation of various types of steel cars at a speed of 40 m.p.h. on track laid

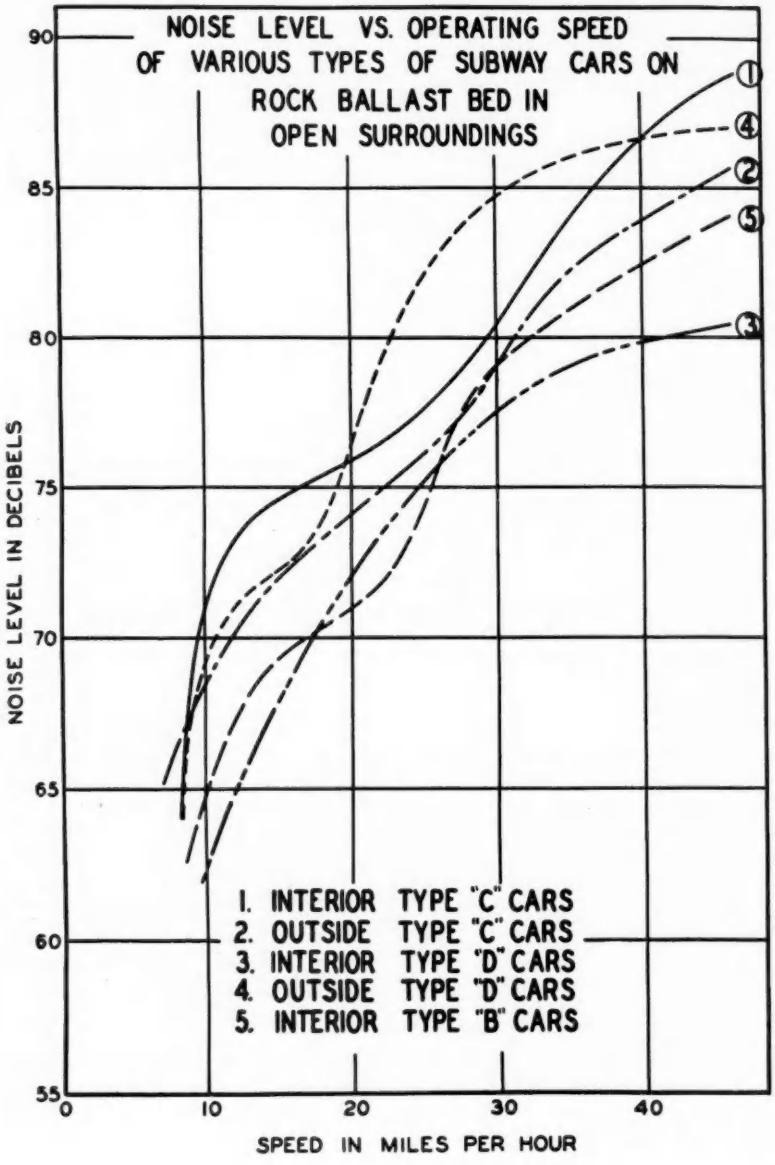


FIG. 8.

on a rock ballasted bed on level ground in substantially open surroundings.

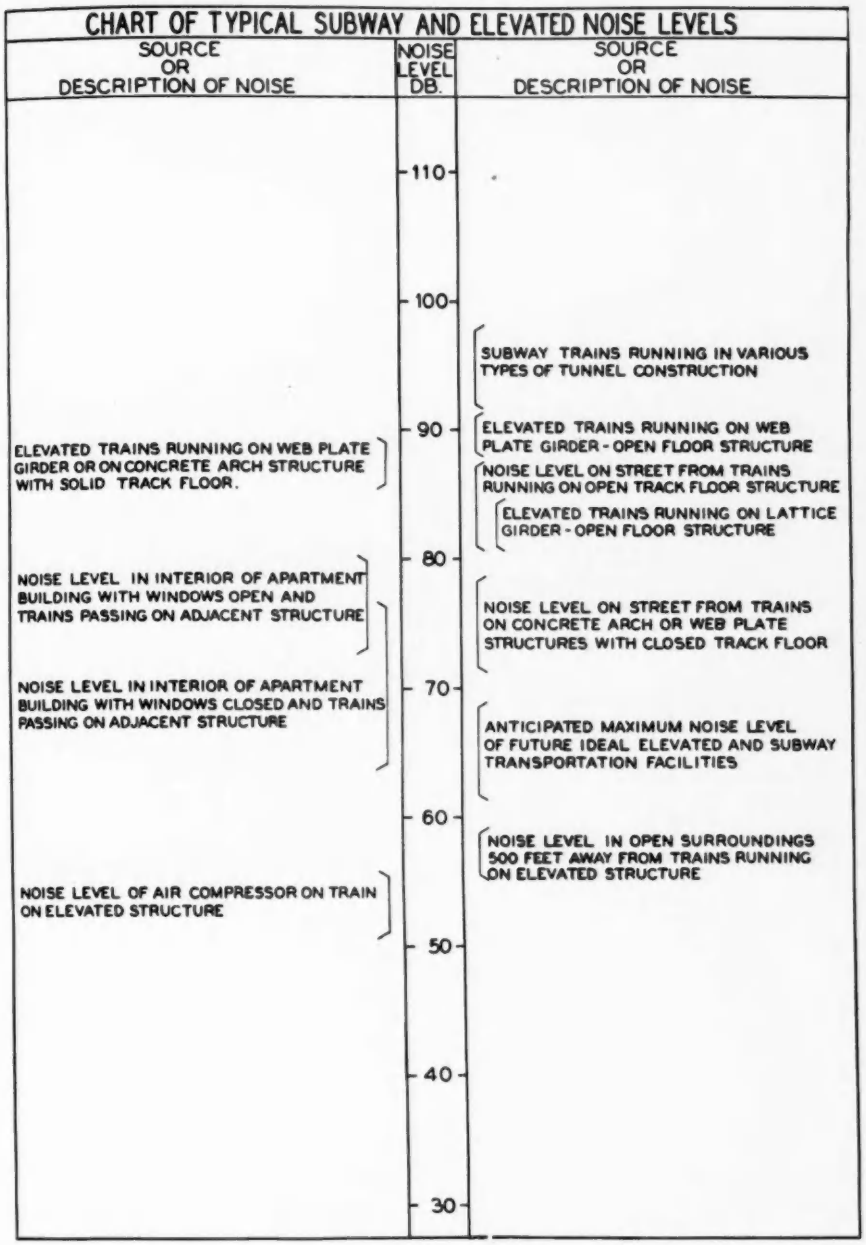


CHART OF TYPICAL SUBWAY AND ELEVATED NOISE LEVELS
SOURCE OR DESCRIPTION OF NOISE
NOISE LEVEL DB.
SOURCE OR DESCRIPTION OF NOISE
110
100
90
80
70
60
50
40
30
SUBWAY TRAINS RUNNING IN VARIOUS TYPES OF TUNNEL CONSTRUCTION
ELEVATED TRAINS RUNNING ON WEB PLATE GIRDER - OPEN FLOOR STRUCTURE
NOISE LEVEL ON STREET FROM TRAINS RUNNING ON OPEN TRACK FLOOR STRUCTURE
ELEVATED TRAINS RUNNING ON LATTICE GIRDER - OPEN FLOOR STRUCTURE
NOISE LEVEL ON STREET FROM TRAINS ON CONCRETE ARCH OR WEB PLATE STRUCTURES WITH CLOSED TRACK FLOOR
ANTICIPATED MAXIMUM NOISE LEVEL OF FUTURE IDEAL ELEVATED AND SUBWAY TRANSPORTATION FACILITIES
NOISE LEVEL IN OPEN SURROUNDINGS 500 FEET AWAY FROM TRAINS RUNNING ON ELEVATED STRUCTURE
ELEVATED TRAINS RUNNING ON WEB PLATE GIRDER OR ON CONCRETE ARCH STRUCTURE WITH SOLID TRACK FLOOR.
NOISE LEVEL IN INTERIOR OF APARTMENT BUILDING WITH WINDOWS OPEN AND TRAINS PASSING ON ADJACENT STRUCTURE
NOISE LEVEL IN INTERIOR OF APARTMENT BUILDING WITH WINDOWS CLOSED AND TRAINS PASSING ON ADJACENT STRUCTURE
NOISE LEVEL OF AIR COMPRESSOR ON TRAIN ON ELEVATED STRUCTURE

FIG. 9.

In Figure 8 it will be seen that the types of steel cars operated under the same conditions described in Table II that each 10 m.p.h. increase

in the operating speed resulted in approximately a 5db. increase in the loudness of the noise created.

In Figure 9 is shown a composite chart of typical elevated and subway noise levels with the observed average variations for different conditions.

It was found that in general maintenance of rail structure could be permitted to vary appreciably without affecting the general noise level created, except for momentary peaks that might be caused by the passage of car wheels over high or low rail joints. It was also found that the maintenance of cars and trucks did influence to an appreciable extent the amount of noise created. The absolute magnitude of this effect cannot be reported since there is no standard criterion of car maintenance.

With respect to subway noise, it is believed that apart from isolated cases of building vibration, the disturbance from this source is primarily confined to the passengers and employees on the line. It was found that in all cases the noise levels were in excess of those measured on elevated structures for equivalent operating conditions. It was found in general that the noise level existing in various types of tunnel construction was a function of the cross-sectional area and the amount of absorption presented by the tunnel walls and surfaces.

TABLE III

AVERAGE NOISE LEVELS IN SUBWAY CONSTRUCTION TRAINS: 10 TYPE "A" CARS SPEED: 35 M.P.H.

Type of Construction	Location of Trains	Average Noise Level
1. Four track rectangular-supporting beams between inside and outside tracks	Inside Track Outside Track	93db. 93
2. Four track rectangular-dividing wall with 2′×7′ openings between inside and outside tracks	Inside Track Outside Track	89 93
3. Three track arched	All Tracks	92
4. Two track rectangular-dividing wall with 2′×7′ openings between tracks	Adjacent Track On other side of wall	95 91
5. Two track semi-circular	All Tracks	96
6. Single track tube	—	97

* All observations in three and four track constructions were in locations adjacent to the outside tracks.

In Table III is presented a summary of the representative average noise levels created by the operation of Type "A" subway car in all types of tunnel constructions. For purposes of comparison, the results have been reduced to the common basis of an operating speed of 35 m.p.h. on similar track construction. The factors affecting noise levels occurring in a tunnel are cross-sectional area and the presence of ventilating and collision walls. The highest noise levels observed were in single track tubes. In the larger four track open tunnels the noise levels were approximately 3db. less than in the two track open tunnels. This difference closely checks analysis and computation. In the Type "A" cars it was found that there was a negligible difference between the interior and exterior noise levels when operated in the tunnels. With the double shell steel cars, it was found that differences of 5db. were obtained for the same conditions except for the type "C" car in which a smaller difference was observed.

The summary of the noise levels prevailing on the existing elevated and subway lines indicates the feasibility of securing major reductions in the amount of noise created. It is anticipated that with the application of basically sound acoustic principles that all of the primary contributory noise sources and their secondary influencing factors may be reduced to a level where the degree of annoyance is no more severe than that experienced due to usual heavy street traffic. It is recognized that a number of new developments must be made before satisfactory conditions will be obtained. In general, it has been found that the trend toward the improvement in the insulation of car bodies indicates the feasibility of obtaining reasonable major reductions of noise in the interior of the cars. This problem involves the construction of car bodies with sufficient sound insulation and vibration isolation to prevent the entrance of exterior noise and the transmission of vibration into the interior of the car. Tests that have been made on the effect of the application of an acoustic absorbing material on the ceiling surfaces of cars show that a reduction of 5db. can be readily obtained in empty cars under normal operating conditions. Calculation shows that this reduction would be equivalent to the acoustic absorption afforded by thirty passengers in an untreated car. However, it is anticipated that the benefits that would be obtained by this method in cars during heavy passenger service would be of but little significance since a greater amount of acoustic absorption would be provided by the passengers themselves. From this may be drawn the conclusion that the trend of improvement in car bodies should be directed toward obtaining greater sound insulation and vibration isolation.

Recent developments in the electric railway field have indicated the possibility of utilizing other types of power drives that are noticeably quieter. In present truck design the motive power is supplied to the wheel axles by spur gear drives which may, or may not, be enclosed. It is common experience that such types of gear drives become extremely noisy and create excessive amounts of vibration due to wear and the inherent characteristics of the gears. The development of worm gear drives with the flexibility permitted by their use, would permit major reductions in the direct amount of air noise created and the transmission of vibration into radiating surfaces of the car body. Some improvement appears feasible in the nature of the gear materials or design and in the use of isolating supports.

Since the major source of noise and vibration is due to impact and friction of car wheels on rails, it is obvious that the development of a resilient wheel will offer major reduction of the primary cause of vibration and noise. Preliminary tests both in this country and France have shown conclusively that resilient wheels offer distinct possibilities in reducing noise conditions created by the operation of transportation facilities. In addition to these developments, it is anticipated that a satisfactory vibration isolation material capable of withstanding the severe service requirements may be used between rail and ties, or ties and structures and be of material aid in reducing some of the disturbing noise conditions produced by the structure. However, it is to be recognized that in any noise reduction program the effect on the overall noise level of reducing one noise source is negligible unless equally effective remedial measures are applied to all contributory noise sources and radiating surfaces. Under many existing conditions, the reduction of vibration in the structure would be of secondary value with the existing cars.

Though it has been suggested that the use of acoustic materials for absorbing air-borne sound would provide an immediately available method of reducing noise conditions, serious consideration cannot be given to this suggestion since it is apparent that there are strict limitations on the amount of reduction that can be secured by this method. A successful program of noise reduction to insure satisfactory conditions can only be obtained through the direct elimination of noise and vibration energy at its source or the prevention of the transmission of vibration to possible radiating surfaces. Measures taken to reduce noise of this magnitude once it is created as an air sound, generally produce a reduction incommensurate with the requirements.

Examining this particular noise problem from its many aspects and

from the results of this survey, it would appear that serious consideration can be given to the continuance and expansion of elevated construction. The cost of usual elevated construction is small compared to the cost of subway lines providing equal facilities. It would be logical and economical, providing satisfactory noise reduction could be positively indicated, to utilize a small part of this tremendous difference in cost of construction, maintenance and operation, to provide noise reduction measures on proposed new elevated lines. It is believed that the most effective and economical noise reduction can be made in new construction compared to the application of remedial measures to existing lines though thorough trials of proposed methods must be made on these existing lines to determine the efficiency of the methods. As the major noise reduction possibilities lie first in improved car design, this benefit would be available to existing lines, and minor improvements applied to existing structures might be justified.

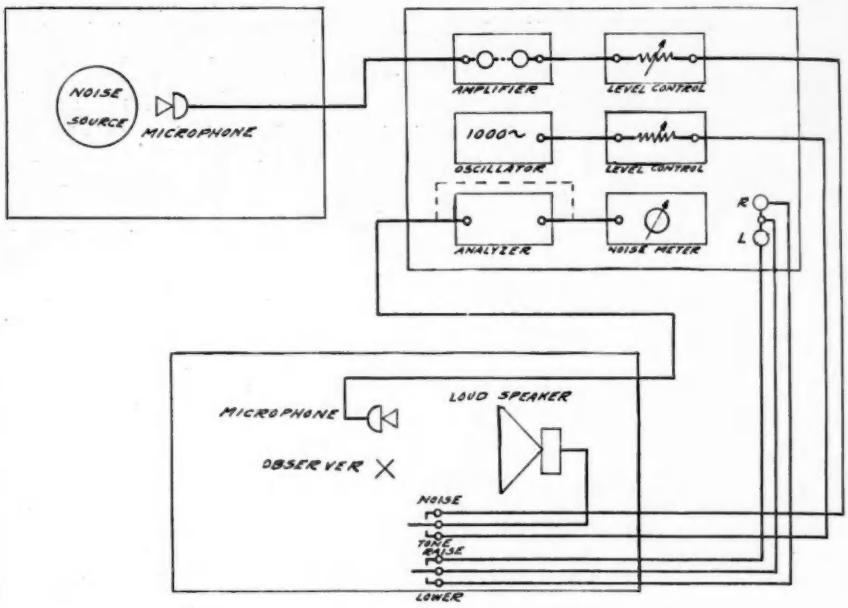
A number of possibilities for the construction and expansion of rapid transit facilities have been indicated that may in some ways solve the noise problems, or, rather reduce the annoyance factor and the resulting effect in depreciating property values. It has been suggested that future regional planning may be so co-ordinated as to permit the establishment of combined rapid transit and vehicular facilities in vicinities where all buildings along the right-of-way have been specially constructed to prevent the transmission of noise into the buildings. In addition, such buildings could act as effective noise shields for property lying outside the zoned area. The possibilities of this suggestion merit serious consideration and may serve as a possible method of solving noise conditions which become impossible of satisfactory reduction.

It has also been suggested that future extension of rapid transit facilities into undeveloped areas may be made using open-cut construction which places the tracks on a level below the surface of the surrounding ground. This type of cut construction produces very noticeable reduction of the range of annoyance caused by the noise. As these areas develop with resulting increases of property values, the objection to the noise conditions will undoubtedly increase. At this time, the construction can be roofed over, making a subway line with possibly a high-speed vehicular highway on the top. Thus, as a community develops, with increasing objections to the prevailing noise conditions, the cost of noise reduction by completing the subway construction can be borne fairly by the community in consideration of the probable increase of property values.

ON THE LOUDNESS OF NOISE

By H. B. Marvin

That the ear is not equally sensitive to all frequencies or pitches of sounds has been known for a long time. Experiments have also shown that the variation of loudness with physical intensity of sounds depends upon the pitch, being least in the neighborhood of 1000 c.p.s.



APPARATUS ARRANGEMENT FOR NOISE TESTS

Fig. 1.

Other effects which complicate the laws of hearing are: masking effects of sounds on each other, the non-linear relation between loundess and intensity, and the integration of components of complex sounds. These are the principal data on which methods of measuring loudness are based.

Nearly all have been obtained with pure tones but it is seldom that the practical determination of loudness deals with pure tones. Not only noise but music also usually consists of a complex mixture of sounds of various intensities and pitches. Consequently it is desirable to test the performance on complex sounds of measuring instruments designed

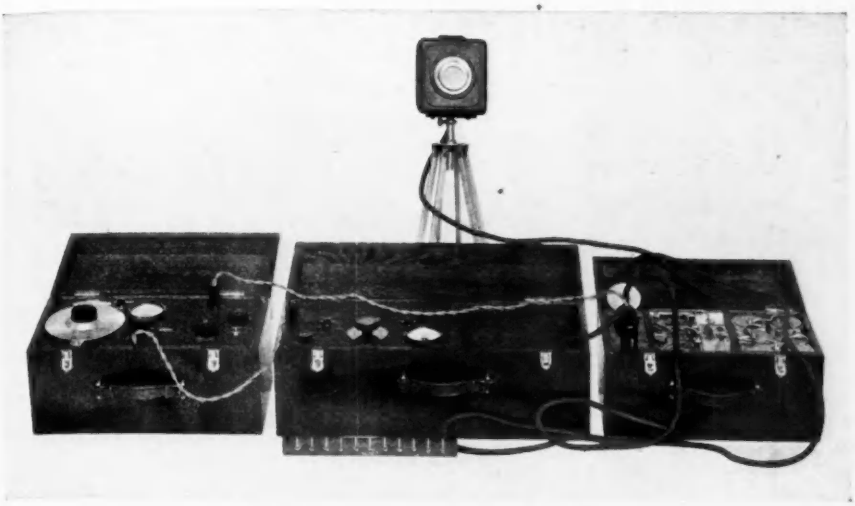
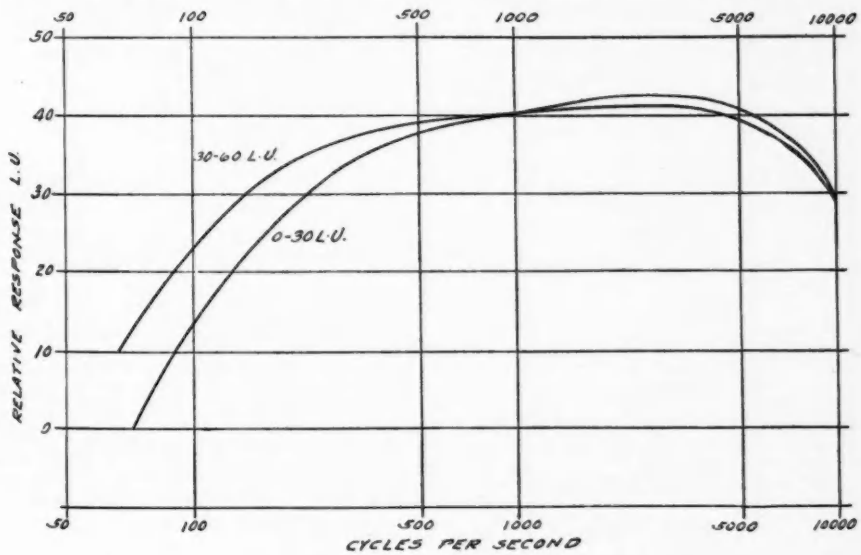


FIG. 2. *Audio noise meter and analyzer.*

from pure tone data. The results of such a test are presented below in preliminary form.

In this test observers were required to adjust the loudness of a noise to equality with that of a pure tone at various levels. Noise and tone were measured with an audio noise meter and results compared.



WEIGHTING CURVES OF NOISE METER

FIG. 3.

The arrangement of apparatus is shown schematically in Fig. 1. The noise sources were located in one room, testing apparatus in another

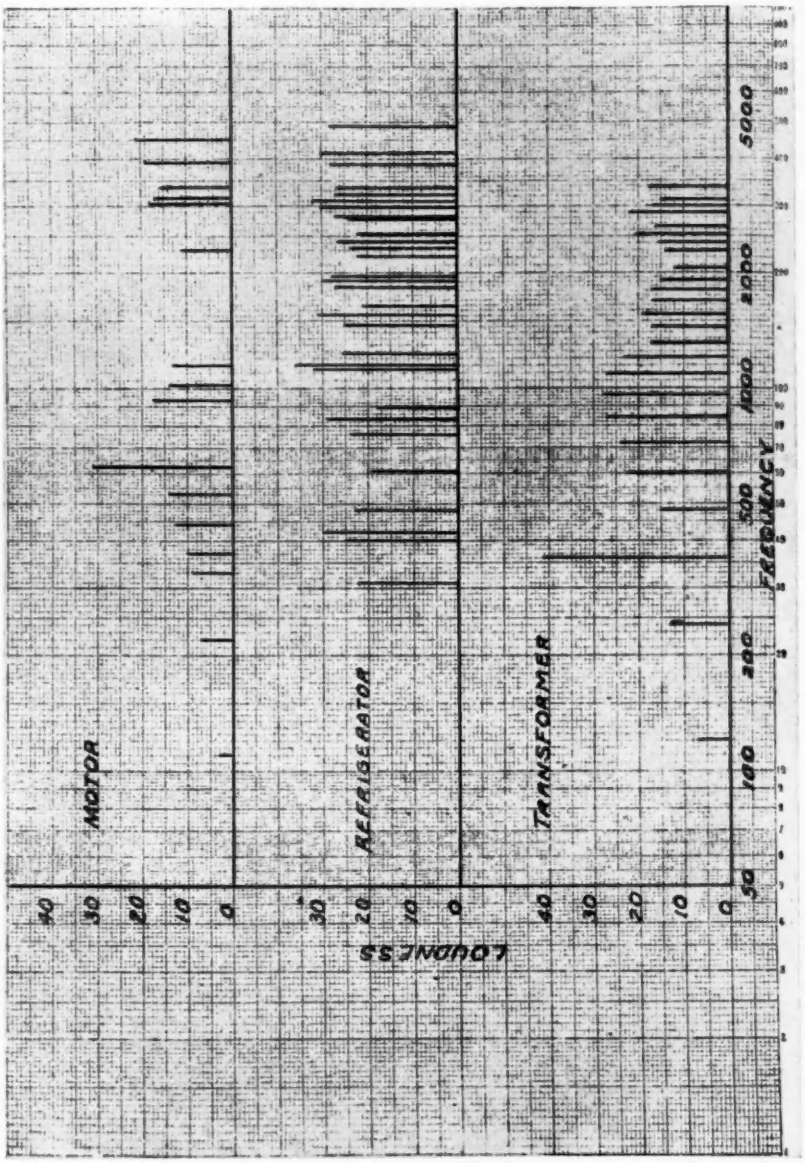


FIG. 4. *Analyses of reproduced noise.*

and the observer in a third. Three kinds of noise sources were used to obtain diversity of character. These noises from a motor, a refrigerator, and a transformer were reproduced in the observer's sound proof room

by a microphone-amplifier-loudspeaker system. Control of loudness of the noise rested with an operator in response to signals from the observer. The same loudspeaker was supplied with 1000 c.p.s. power from a beat frequency oscillator, the power level being set in the control room. The observer was able to switch the loudspeaker from noise to tone and to signal for variations of noise level by means of two keys. The microphone of the audio noise meter was located beside the observer but the amplifier and analyzer were in the control room.

This noise meter, Fig. 2, is similar to one previously described[1] except for the following modifications:

Reference frequency is 1000 c.p.s. and reference sound pressure is .001 bar. The scale is calibrated in "loudness units" which are numeri-

1000 C.P.S. TONE LEVEL	17	22	27	36	44	59	74	88.5
MEAN OF NOISE METER RDGS.	18.5	23	27.5	36.5	42.5	55	69	82
METER ERROR	+1.5	+1	+0.5	+0.5	-1.5	-4	-5	-6.5
PROBABLE ERROR OF BALANCE	1.2	2.3	3.7	5.0	6.7	6.2	6.0	5.8

RESULTS OF NOISE TESTS

FIG. 5.

cally equal to decibels above the reference point. The meter contains two weighting networks one of which is used for readings below 30 and the other above 30 loudness units. These networks give the meter the response characteristics shown in Fig. 3 and are close approximations to constant loudness curves through points which at 1000 c.p.s. are respectively 15 and 45 loudness units above the reference point. The curves were computed by interpolation from Kingsbury's data.[2]

The character of the noises used in the test is described by the analyses of Fig. 4 which were made with the audio noise meter. These represent reproduced noise at the observer's location. In the case of the transformer all components are harmonics of 120 cycles, while in the other two there is the usual mixture.

Comparisons at each tone level were made, using 8 observers. In the table, Fig. 5, the test results are given, in loudness units above 1 millibar. Each figure is the mean of 16 observations. The difference between

[1] "The Measurement of Machinery Noise" by H. B. Marvin. Electrical Engineering, Vol. 50, p. 349, May 1931.

[2] "A Direct Comparison of the Loudness of Pure Tones" by B. A. Kingsbury. Physical Review. Vol. 29, p. 588, 1927.

noise meter reading and tone level, called meter error includes all the errors of the system.

These contain among others:

(1) Error of mean due to small number of observations.
(2) Variations of noise itself.
(3) Approximations in weighting networks.
(4) Errors due to RMS integration.
(5) Errors due to masking.

The probable error of balance means that in a single observation it is equally probable that the result differs from the mean by more or less than this amount.

From the fact that the agreement is good in the range well covered by the weighting networks it is expected that a third network to be used in the upper part of the range will improve the accuracy.

In view of the complexity of the ear characteristics it is encouraging to find that relatively simple instruments designed from data obtained with pure tones are capable of measuring noise in substantial agreement with subjective methods.

THE APPARENT REDUCTION OF LOUDNESS

By Donald A. Laird, Emery Taylor, and Herman H. Wille, Jr.*

Colgate Psychological Laboratory, Hamilton, New York

Translating decibel levels into the daily experience of the average person is perhaps the most difficult task faced by the acoustical engineer. When a considerable amount of money is to be appropriated for making a work place quieter, for instance, the engineer can say that after acoustical material is added the noise level will be reduced by five or ten decibels. "But how much quieter will that make the office," is likely to be the inquiry. "A great deal" is not only an unsatisfactory but an unscientific answer. This work was undertaken in the attempt to give a basis for a scientific answer to such inquiries in a quantative fashion based upon human experience and human judgment.

Aside from this major and practical aim of the work, the findings raise many important theoretical problems which concern the Weber-Fechner law, question the desirability of using the term "sensation unit" interchangeably with decibel, and possibly lead to the suggestion that a new approach be made to the calibration of the audiometer for some phases of acoustical work. These theoretical questions are being reserved for later discussion.

Parkinson has previously reported to the Acoustical Society of America observations which indicate an illusory impression of a great reduction in loudness when the actual decrease in decibel level is rather small. Conversation with other workers in the field in general confirm this observation.

Method of Experimentation

Observers were given a buzz from the 3-A audiometer for three seconds. A three second interval of relative silence followed this *original level*, and after this a less intense buzz was given for another three seconds. The observer then reported whether the second buzz was half as loud as the original level, or whether it needed to be increased or decreased in loudness to be half as loud as the original. This procedure was repeated, if necessary, until a second level was found which the ob-

* With the assistance of Robert Barnum, Allan Beall, Morford Downes, Edward Downey, Cameron Farquhar, Lorenz Franz, Robert Hildreth, Jordan Hunt, Richard Jones, Donald Keeler, Edward Khachmanian, Charles Morell, Mellie Mooshie, Richard Rahn, Robert Sale, Donald St. Clair, Henry Treadwell, Ross Weeks. (Courtesy The Celotex Co.)

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server reported as being half as loud as the original. The second buzz of each pair was varied in loudness in haphazard order to avoid an error of expectancy.

A similar procedure was followed later to find the decibel levels at which the original level appeared reduced by one-fourth as well as reduced by three-fourths.

All observers were young men and in good health. Only those who had normal thresholds of hearing were used.

They were paid forty-cents an hour while taking part in the experment.

The observers were naive and not familiar with the calibration of the audiometers. This was found essential in preliminary work where those well acquainted with the audiometers and who had done considerable work in measuring noise levels were used. Erratic and misleading results were obtained from these acoustically sophisticated observers, and for an interesting reason. When the original level was sounded they would instantly identify it—with approximate accuracy—as being 60 decibels. Following this they would listen until a second stimulus was given which they estimated to be at the 30 decibel level. In other words, it was not possible for them to rule out their previous experience in identifying decibel levels and respond primarily to loudness as such.

APPARENT REDUCTION IN LOUDNESS OF NOISE HEARD IN ONE EAR

Reduction of Loudness by One-half: Table 1 gives the data for the apparent reduction in loudness by one-half. The average of this table is

TABLE I

REDUCTION IN DECIBEL LEVEL LOUDNESS NEEDED TO REDUCE THE LOUDNESS OF THE 3-A AUDIOMETER BY HALF

Decibel level to be "Halved"	Level which appears to halve the original, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
100	70	75	80	60	80	80	80	75	70	70	74
90	65	70	75	60	75	75	65	70	65	65	68.5
80	60	70	60	60	55	60	50	65	55	60	59.5
70	50	60	55	50	50	50	45	60	45	50	51.5
60	35	50	45	40	40	35	40	40	40	45	41
50	30	40	35	30	30	30	25	40	30	35	32.5
40	25	30	25	25	30	15	20	35	20	20	24.5
30	20	25	15	15	20	15	15	20	10	15	17
20		15	10		15	10	10		5	10	10.7
10											

plotted as the middle curve in Figure 1. A loudness of 80 decibels appears to be reduced one-half when it is reduced only to 59.5 decibels rather than to the actual half level of 40 decibels.

A slight reduction appears like a reduction of approximately twice the actual decrease in decibels—at the "louder" levels; as the quieter

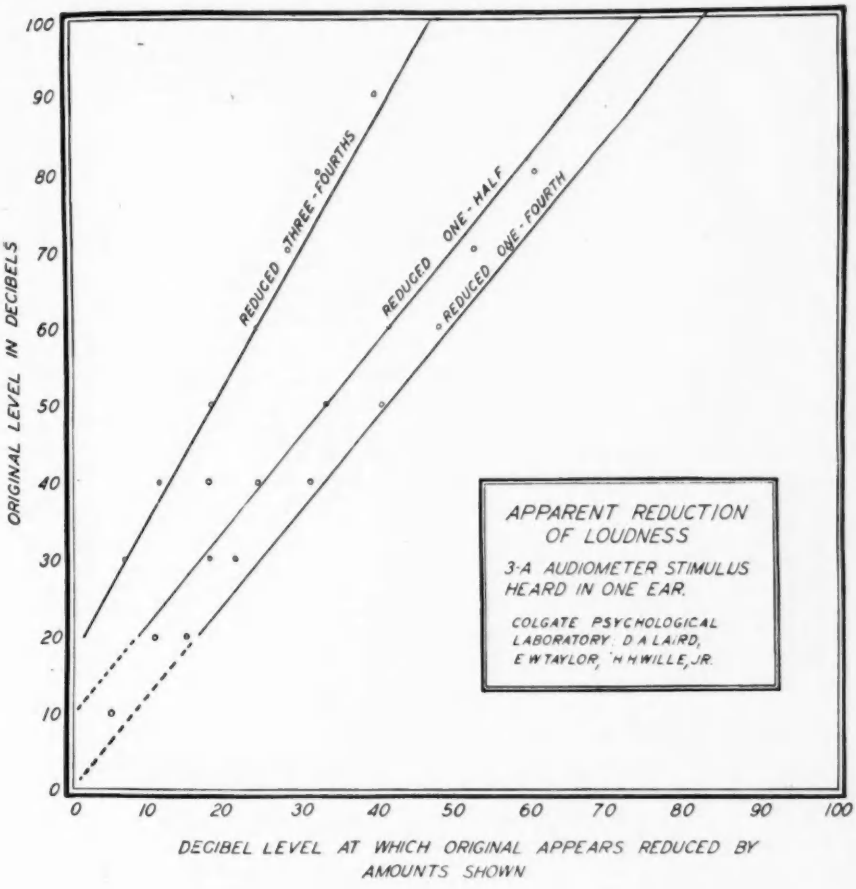
ORIGINAL LEVEL IN DECIBELS
DECIBEL LEVEL AT WHICH ORIGINAL APPEARS REDUCED BY AMOUNTS SHOWN
REDUCED THREE-FOURTHS
REDUCED ONE-HALF
REDUCED ONE-FOURTH
APPARENT REDUCTION OF LOUDNESS
3-A AUDIOMETER STIMULUS HEARD IN ONE EAR.
COLGATE PSYCHOLOGICAL LABORATORY. D. A. LAIRD, E. W. TAYLOR, H. H. WILLE, JR.

FIG. 1.

levels are approached the reduction to appear as fifty per cent reduction comes nearer to a mathematical halving of the original buzz. This is shown clearly in the tabulations of Table 2, the data for which were calculated from Table 1.

Verification by "doubling": The findings to this point were so striking that verification was sought by attempting to work the reverse. In this phase of the experimentation the same observers were asked to select

TABLE II

THE INCREMENTS OF CHANGE TO PRODUCE THE IMPRESSION OF HALVING ARE NOT UNIFORM, BUT VARY SOMEWHAT CONSISTENTLY WITH THE LOUDNESS

Decibel level	Decrease to produce "halving"	
	In decibels	In per cent change
80	20.5	25
70	18.5	26
60	19	32
50	17.5	35
40	15.5	39
30	13	43

the second buzz which appeared to double the loudness of the original level. The data on doubling verified the observations on halving to a remarkable degree. A level of 80 decibels, for instance, appears to be halved when reduced to 59.5 decibels, and 60 decibels appears doubled by 79 decibels. Similarily 60 decibels is halved by 41, and 40 is doubled by 59.5; while 50 decibels is halved by 32.5, and 30 is doubled by 50. The full data are in Table 3.

TABLE III

INCREASE IN DECIBEL LEVEL NEEDED TO DOUBLE THE APPARENT LOUDNESS OF THE 3-A AUDIOMETER

Decibel level to be "Doubled"	Level which appears to double the original, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
80	100	90	90	100	90	100	90	95	95	95	94.5
70	90	80	80	80	80	80	90	85	90	85	85
60	80	70	80	85	75	75	85	75	80	80	79
50	70	60	65	75	65	70	70	60	70	70	67.5
40	65	60	60	65	55	60	60	50	65	55	59.5
30	45	50	50	55	50	55	50	40	45	50	50
20	35	40	40	40	40	25	40	35	35	40	36
10		25	25	25	25	25	25			20	24.3

There is some slight disagreement between halving and doubling. This is not greater than would be due to the experimental error, and the observations on doubling confirm those on halving.

Reduction of loudness by one-fourth and by three-fourths: The data for these are given in Tables 4 and 5, and are included in Figure 1. These reductions follow in general those for halving and a tentative law may be

TABLE IV

APPARENT REDUCTION OF LOUDNESS BY ONE-FOURTH

Decibel level to be reduced	Level which appears to reduce the original one-fourth, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
100	80	80	85	90	85	75	70	75	80	80	80.7
90	75	75	75	75	80	75	75	65	75	75	75
80	70	70	75	70	75	70	70	55	70	70	70
70	55	55	60	60	60	55	60	50	60	60	57.8
60	45	50	45	45	50	45	50	40	50	50	47
50	40	35	40	40	40	40	45	35	40	45	40.7
40	30	30	30	30	30	30	35	25	30	35	31.4
30	25	20	20	20	25	20	20	15	20	25	21.4
20	15	15	10	15	15	15	15	5	15	15	15
10	5		5	5	5	5			5	5	5

proposed to the effect that *for a level of 80 decibels the loudness appears to be reduced twice as much as the percentage reduction in actual decibel level, while at the 30 decibel level the percentage reduction in actual level and the apparent reduction in loudness are the same, with an even gradation in the "illusory increment" between these extremes.*

TABLE V

APPARENT REDUCTION OF LOUDNESS BY THREE-FOURTHS

Decibel level to be reduced	Level which appears to reduce original by three-fourths, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
100	45	55	50	55	55	45	50	50	50	45	49.3
90	35	45	45	40	45	35	45	45	45	35	40
80	35	30	30	35	30	35	35	35	35	30	32.8
70	30	30	30	30	30	25	30	30	30	25	28.6
60	25	20	20	25	25	20	25	25	25	30	24.3
50	20	20	15	15	15	15	20	20	20	20	18.6
40	15	10	10	10	10	10	15	15	10	15	11.8
30	10	5	5		5	5		10	5	10	7
20						5			5	5	
10											

Independence from pitch: Previous work in this laboratory has shown that annoyance caused by noise bears a relation to pitch as well as to loudness. A check was therefore attempted on the general applicability of the data in Figure 1 by another experiment in which the relatively

pure pitches of the 2-A audiometer were used, rather than the complex buzz of the 3-A audiometer which was used in the work reported thus far. The data from the buzz-type audiometer are, of course, more important in practical work since it simulates actual noises which are ordinarily bothersome. The work on the different pitches is primarily of theoretical significance.

TABLE VI

APPARENT REDUCTION OF LOUDNESS OF PITCH OF 4096 CYCLES BY HALF

Decibel level to be "Halved"	Level which appears to reduce original by half, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
80	60	65	70	60	65	60	60	60	60	60	62
70	45	55	55	50	55	50	50	50	50	50	51
60	35	45	40	35	45	35	35	35	40	40	38.5
50	25	35	35	25	30	30	25	25	30	35	29.5
40	20	30	20	20	20	20	20	15	20	20	20.5
30	15	20	15	10	15	15	10	15	10	15	13.0
20		10	10	5	10		5			10	7.1
10											

TABLE VII

APPARENT REDUCTION OF LOUDNESS OF PITCH OF 1024 CYCLES BY HALF

Decibel level to be "Halved"	Level which appears to reduce original by half, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
100	65	90	80	75	80	75	70	75	75	75	76
90	65	80	70	65	70	65	65	65	65	70	68
80	50	65	65	65	65	55	60	60	55	60	60
70	45	55	50	50	55	45	45	50	50	50	49.5
60	35	45	40	40	45	40	40	40	40	40	40.5
50	30	35	30	30	35	30	30	30	30	30	31
40	20	25	20	20	20	20	20	25	20	20	21
30	15	15	15	10	15	15	15	20	10	15	14.9
20	5	10	5	5	10	10	5	5	5	5	6.5
10	5			5							5

The data for the separate pitches are given in Tables 6, 7, 8 and 9. These agree remarkably with the data for the buzz noise. Obviously the apparent reduction in loudness is not dependent upon the pitch of the noise. The illusory increments are practically identical for all of the pitches and for the buzz type stimulus.

TABLE VIII

APPARENT REDUCTION OF LOUDNESS OF PITCH OF 256 CYCLES BY HALF

Decibel level to be "Halved"	Level which appears to reduce original by half, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
100											
90											
80	50	70	65	70	60	60	55	55	60	65	61
70	45	60	55	50	55	55	50	45	50	55	52
60	35	50	45	40	50	45	40	40	40	40	42.5
50	25	35	35	25	40	35	30	30	30	35	32
40	20	30	20	15	30	25	20	25	20	20	22.5
30	10	20	20	10	25	20	15	10	20	15	16.5
20	5	10	10	5	10	10	10	10	10	10	9
10		5									

TABLE IX

APPARENT REDUCTION OF LOUDNESS OF PITCH OF 64 CYCLES BY HALF

Decibel level to be "Halved"	Level which appears to halve the original, in decibels										
	Observers										
	R.B.	A.B.	M.D.	C.F.	D.K.	C.M.	E.T.	H.T.	R.W.	H.W.	Avg.
100											
90											
80											
70											
60											
50	20	40	35	25	35	30	30	25	30	30	30
40	20	30	25	20	30	25	20	25	25	25	24.5
30	15	20	20	15	20	20	20	20	20	20	19
20	5	10	5	5	10	10	10	10	5	10	8
10					5					0	2.5

APPARENT REDUCTION IN LOUDNESS OF NOISE BATHING THE OBSERVER

In actual operating conditions the noise bathes observers, striking both ears rather than entering just one ear as was the case with the observations reported thus far in which the head-band receiver of the audiometer was at one ear of the subject. Further observations were therefore made in which the audiometer outut was passed through a special amplifier so that it filled a moderately absorbent room and bathed the subject. The impedances of the amplifier were designed to match the

audiometer in order to prevent distortion. Zero decibel level was empirically determined and the potentiometer scale of the audiometer used thereafter.

The data for this phase of the work are charted in Figure 2. Ten observers were used. This is in close agreement with the corresponding

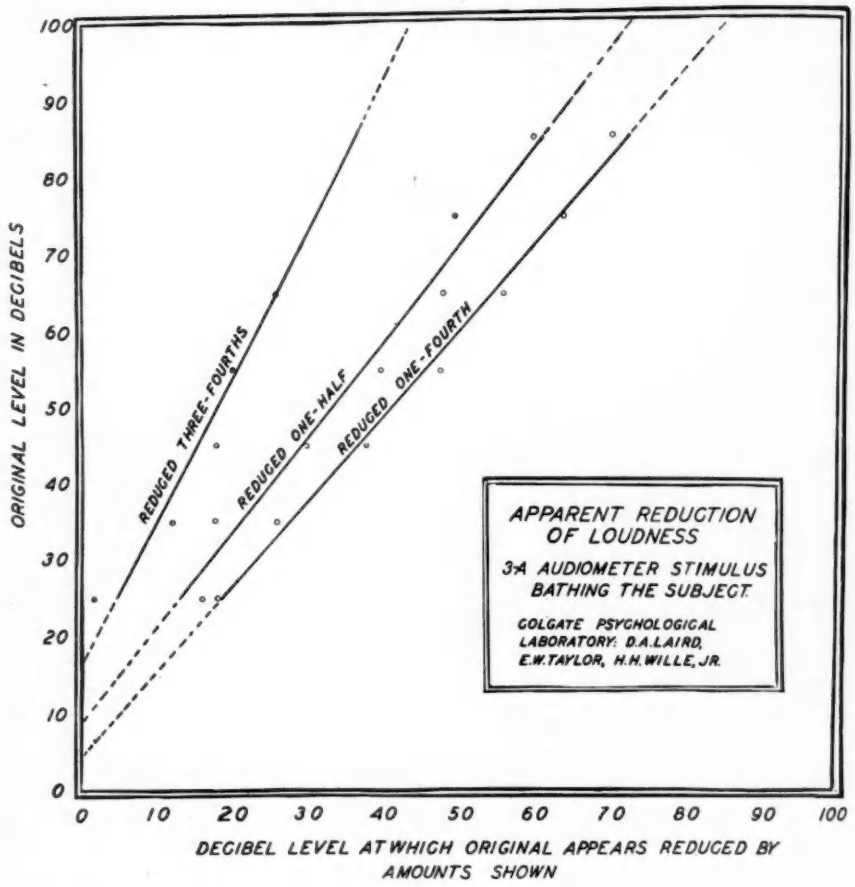

FIG. 2.

data for the stimulus being heard in one ear only. Since both figures are practically indentical it is plain that the amount of apparent reduction is independent of whether or not the noise bathes the observer or is heard only in one ear.

CLOSE AGREEMENT AMONG OBSERVERS

In psychological measurements individual differences are the rule unless one is measuring a trait of a rather basic biological quality. Ex-

amination of the detail tables of data will reveal no marked individual differences. Whatever the mental components may be, the observers who were without opportunity for collusion agree closely upon what reduction appears to be one-fourth, one-half, or three-fourths. Their agreement, in fact, is closer than is obtained for selecting lifted weights which appear to be half the weight of a standard which is being tested against.

Not Due to Memory Fading

We have considered the possibility that the findings might be complicated by a fading of the memory image of the original stimulus level. It will be recalled that the observers were not familiar with the nomenclature or range of decibels on the audiometer dials as well as having not previously listened closely to compare or contrast sounds at all critically; this naïvette might conceivably cause a rapid fading of the memory image of the stimulus to be reduced.

The test procedure was adapted to test this possibility. For this test the observers were to report whether or not the second buzz was the same loudness as the original buzz of the pair. A fading of the memory image by 5 decibels would result in a second buzz 5 decibels lower than the original being identified as the same as the first. Without exception all observers made the correct identifications.

The findings are not due to a fading of the memory image and we are apparently dealing with a mental concept of reduction that is stable, uncomplicated by individual differences to any extent, and which is very generous since a slight reduction appears remarkably large.

TRANSMISSION OF SOUND THROUGH APERTURES

By Earland Ritchie

Indiana State Teachers College, Terre Haute, Indiana

Abstract*

This investigation was an attempt to obtain experimental data to justify a prediction made by Lamb concerning the fractional transmission of sound through an aperture in a very thin plate of approximately infinite boundaries. The effects upon the actual transmission of varying the lengths of the apertures, of changing the number of the apertures, and of increasing the aperture sizes were investigated.

The work was done in the Sound Absorbing Acoustic Laboratory since it approximates outdoor conditions. The purest tones available were used as sources of sound. The apertures tested varied in lengths from one thirty-second of an inch to twelve inches, and in diameters from one-fourth of an inch to one and three-fourths inches. These were circular apertures cut through large plates three feet square. Equations were established to express the observed relationships found existing for the lengths of apertures, and for a variation of the number of apertures.

The theory given by Lamb for plates of infinitesimal thicknesses was found to be in agreement with experimental results. Apertures of different lengths showed an exponential relationship when compared with the per cents of transmission. The greatest transmissivity of an aperture can be obtained by using one aperture only. Increasing the number of apertures gave an ever diminishing result per aperture. The transmission through apertures of any length is directly proportional to their respective cross-sectional areas.

* (This work was done at Indiana University with the aid of Dr. A. L. Foley and Dr. R. R. Ramsey.)

The purpose of this study was to obtain, if possible; (1) experimental evidence for some statements made by Lamb concerning transmission of sound through an aperture, (2) to find the effects upon the actual transmissions, of the depth or length of apertures in plates of different thicknesses, (3) of gradually increasing the number of apertures, and (4) of the size of the apertures.

The theory of the transmission of sound through an aperture was early treated by Rayleigh[1], and a little later by Lamb[2]. Lamb[3] states that for circular apertures the conductivity (K), which is the nature of a length, is equal to "2a" where "a" is the radius of the aperture. He then writes that it appears for a circular aperture that the transmission is $8/\pi^2$ or .816 times the energy propagated across an equal area (πa^2) in the primary waves.

Stewart and Lindsay[4] write that no aperture is ever in a wall of infinitesimal thickness as was assumed by Lamb in his statement con-

[1] Rayleigh, Theory of Sound, Vol. II, p. 173.
[2] Lamb, Dynamical Theory of Sound, p. 244.
[3] Lamb, Dynamical Theory of Sound, p. 246.
[4] Stewart and Lindsay, Acoustics, p. 53.

cerning the transmission of an aperture. Therefore they say it is necessary to consider the influence of the thickness of the wall. There is then involved a channel and an orifice effect. The conductivity is given to be equal to $\pi\alpha^2/1-2\alpha$, where $\alpha=(\pi/4)\ a$ which is in general too small. Rayleigh[5] shows that "α" varies between $\pi a/4$ and $8a/3\pi$ as inferior and superior approximate limits.

Foley[6,7] has found that the velocity of sound in tubes is less than in free air. He also showed that the speed depends upon the intensity of the pulse entering the tube; that the velocity decreased with the diameter of the tube; and that the higher the frequency the less the retardation. The material of the walls of the tube more or less affects the velocity of sound.

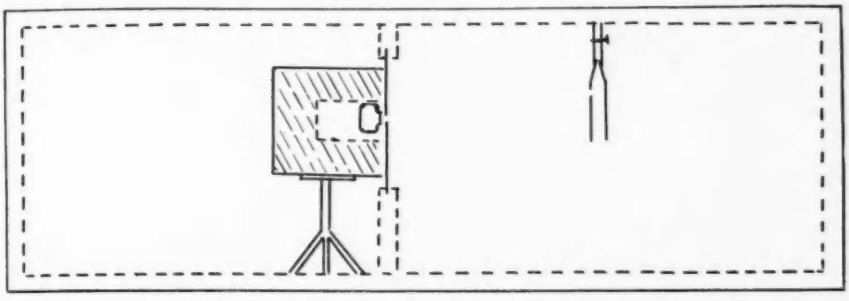

FIG. 1. *A cross-section diagram of the laboratory showing the source room on the right, receiving room on the left, and the apparatus in working positions.*

Experimental work on sound transmission would more nearly approach normal conditions if multiple reflections and reverberations were minimized considerably. Under such conditions a source and a receiver could be moved about in the rooms without being so greatly affected by the room constants. The Acoustic Laboratory of Indiana University described by Foley[1] proved to satisfy best the required conditions. The partition separating the source room from the receiving room was modified for this particular problem.

The door and window openings, when closed, were made as nearly sound proof as possible. The door and plate in the window were readily demountable to permit adjustments, open window readings, and exchange of transmitting plates in the window.

[5] Rayleigh, Theory of Sound, Sec. 307.

[6] Foley, Proceedings Indiana Academy of Sciences, 1919, "The Velocity of Sound Waves in Tubes."

[7] Foley, Physical Review, Aug. 1919, "The Speed of Sound Pulses in Pipes."

[1] Foley, Proceedings of Indiana Academy of Science, 1924.

The microphone unit was the only piece of apparatus used in the receiving room. In order to avoid effects of feeble reflected waves or even faint sound patterns the unit was placed inside a cubical box about three feet on each edge, and which was lined on all sides and back with six thicknesses of balsam wool. The box was then placed on a tripod stand thus making the entire assemblage portable so that it could be placed at various positions in the room. The microphone front was kept parallel with the plate containing the apertures and on an axis through the aperture. The transmitter distances from the plate varied from one-fourth inch to approximately four feet.

The source room conditions were kept constant. Sources of sound used were organ pipes, and constant frequency records with sound from a square horn. The frequencies of the organ pipes used were UT_4, FA, and UT_3 respectively. Data were obtained by using the pipes swinging through a constant arc. The pipes were swung in a plane parallel with the plates in the window and were propelled by a phonograph motor. They were worked at a pressure of 10.5 cms. of water. The constant frequency records were used primarily to obtain a higher frequency than was available in organ pipes. They also proved to be giving more pure tones while the organ pipes always had overtones present. The limit of higher frequencies used was around 1800 v.p.s. This limitation was due to the cut-off of the transformer in the amplifier of the receiver.

The receiving apparatus used consisted of a Jenkins and Adair type condenser transmitter unit, a three stage amplifier, a step down transformer supplying a thermo-junction, and a microammeter giving the responses by its deflections. All amplification factors were kept constant.

A calibration of the thermo-junction made possible the conversion of all micro-ammeter responses into amplifier output currents expressed in milli-amperes squared.

The relationship existing between the intensity of the sound impulse upon the condenser diaphragm and the amplifier output was assumed to be that of a straight line when plotted.

I. Transmission Through Apertures of Various Lengths

In order that the transmission of sound through apertures of various lengths might be investigated, materials of different thicknesses were used as plates in the partition window of the laboratory. The plates were tin, box-board, and Nu-wood. The tin plate was one thirty-second of an inch in thickness, and the box-board plates were one-sixteenth, three-sixteenths, and three-eighths inches respectively. The thicknesses

of the Nu-wood plates were one-half, eleven sixteenths, seven-eighths, and one inches.

It was found that consistent and reliable data could be obtained only by using as pure a tone as possible. For this reason constant frequency records were used with the reproducing horn at a distance of approximately forty-eight inches from the front of the apertures. Wente[1] has shown that if the sound impulses are very intense the transmitter response is distorted. By increasing the sound source distance the effective response was lessened. This distance could not be made too great and still approximate a plane wave front from the reproducer horn. The horn front was twenty inches square.

The microphone had an effective diaphragm exposure of one and three-fourths inches in diameter, so that in an open window reading it was responding to an area of sound impulses one and three-fourths inches in diameter. The diameters of the apertures were made the same size as that of the transmitter diaphragm, thus permitting calculations of the fractions of sound transmission.

The transmitter was placed as close behind the aperture as possible without being in contact with the plate. This position was used for several reasons; the transmitter would receive practically all the transmitted waves before they were much dispersed since the aperture acts as a point source according to the Huygens principle. Little or no transmission of sound energy by the plate could enter the microphone face. When the plate was removed the transmitter was practically in the same position as was the aperture previously. Open window readings could then be taken.

Figure 2 shows the curves resulting from plotting per cents of transmission against the aperture lengths in inches. The per cents of transmission were obtained by the ratio of the transmission values of the aperture and open window when expressed in millamperes squared.

The curves are of the exponential form and can be fitted by the equation $y=be^{ax}$. The resulting value of "a" from 1500 frequency curve is $-.3108$. The value of "b" became 7.8585×10^{9}. Solving for "x", which is the per cent of transmission we get $X=K \log y-D$, where "y" is plate thickness in inches. The expression for the 1500 v.p.s. curve becomes $x=73.308-7.4084 \log y$. The constants in the equation $y=be^{ax}$ depend upon the units used.

The greater discrepancies present among the lower frequencies were

[1] Wente, E. C., Physical Review, July 1917, pp. 39–63. "A Condenser Transmitter as a Uniformly Sensitive Instrument for Absolute Measurement of Sound Intensity."

attributed to the presence of sound patterns in the source room; comparatively greater intensity of worn record and reproducer noises than the desired constant frequency source; and possible slight influencing

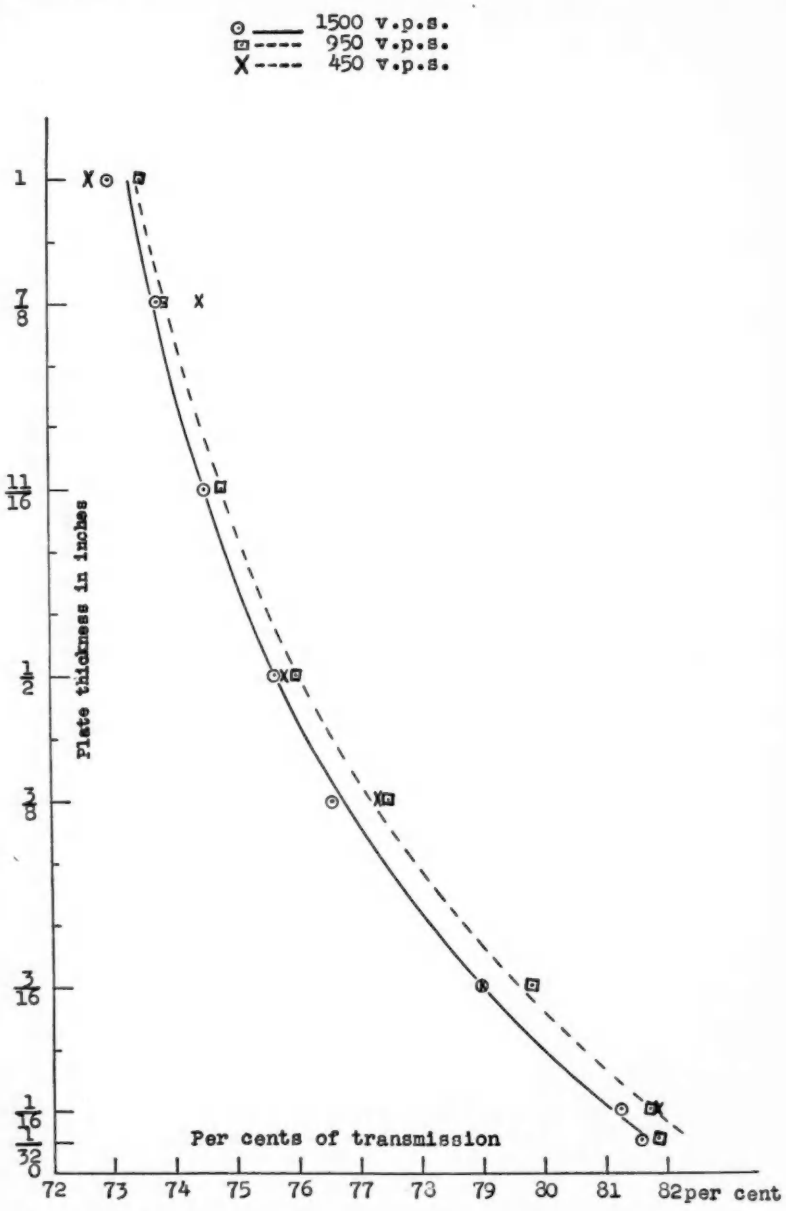

FIG. 2. *Transmission curves relative to open window values showing the effects of the length of apertures.*

effects of the body vibrations of plates which would be greater for lower frequencies.

The results herein given pertain to apertures one and three-fourths inches in diameter in plates up to one inch in thickness.

The fractions of transmission for the thinnest plate used were; .816, .818, and .816. The value given by Lamb was .816 for an infinitesimally thin plate. According to the exponential relationship which proved to be the situation found in this study the decrease in fractional transmission was greatest for thinner plates or sheets, and decreased more slowly as the thicknesses became greater. The values found for one inch plates were: .727, .735, and .730. For plates or walls of several inches or even feet in thickness the values according to the curves given herein would still be .700 or above. At such thicknesses, however, numerous other factors enter so that it is doubtful if the above named relationship would accurately hold. Such conditions in a practical sense became transmission through pipes and not transmission through apertures in plates or walls.

Different upper limitations might be experienced for apertures of other diameters. There is all reason to believe that all sizes of apertures in infinitesimally thin plates would have the same lower limit of fractional transmission.

II. Transmission Through Different Numbers of Apertures

The transmission of sound through a number of adjacent apertures was worked with the number of apertures beginning with one and gradually increased by one until a limit in number was established by the perpendicularity of the incident wave fronts. The limiting number attempted was twelve apertures arranged rectangularly in placement and spaced three-fourths of an inch from center to center. The sizes used were one-fourth, three-eighths, and one-half inches in diameter.

The circular apertures were cut in a steel plate fitted into the partition window. The plate was one-sixteenth of an inch in thickness and was covered, except for a small region about the apertures, by a sheet of balsam wool. This covering prevented intense reflections from the steel plate and also reduced the amount of transmission through and by the plate itself because of the absorption in the wool.

The transmitter was placed at distances of two and one-half, eight, and twenty-two and one-fourth inches back of the plate.

The sound sources were swinging organ pipes of frequencies UT_3, FA, and UT_4 placed at distances of eighteen inches and seventy-two inches

from the front of the plate. The varied source distances changed the intensities of response. When the source was at a distance equal to any multiple of a half wave length the aperture came at a node of the room pattern and so the transmitted intensity was greater.

When the actual transmissions were placed on a relative basis and plotted against the number of apertures the same type of curve resulted in every case. Since most of the readings came on the straight part of

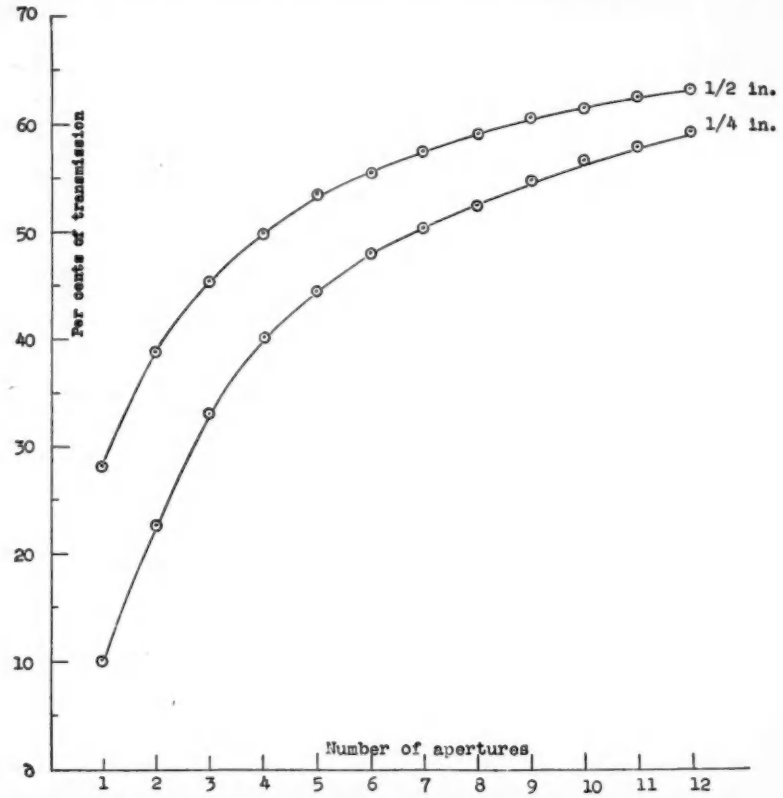

FIG. 3. *Transmission curves relative to open window values showing the effect of the number of apertures. Aperture diameters ¼ and ½ inches.*

the thermo-junction calibration curve the same form curve resulted when millamperes squared were plotted against the number of apertures. Also the transmitted values relative to the open window reading gave the same form of curve since the open window reading was a common divisor for any one set of conditions. Most of the curves given are in terms of the relative per cents of transmission as compared with the transmission of one aperture.

Each of the plotted values is the average of eight repeated readings. The closed plate reading was obtained, and then in turn the number of apertures was increased as readings were made. Readings were made so that the average is an approximate integrated result with respect to position.

The calculated per cents of transmission with respect to the open window value were plotted against the number of apertures as is shown in Figure 3.

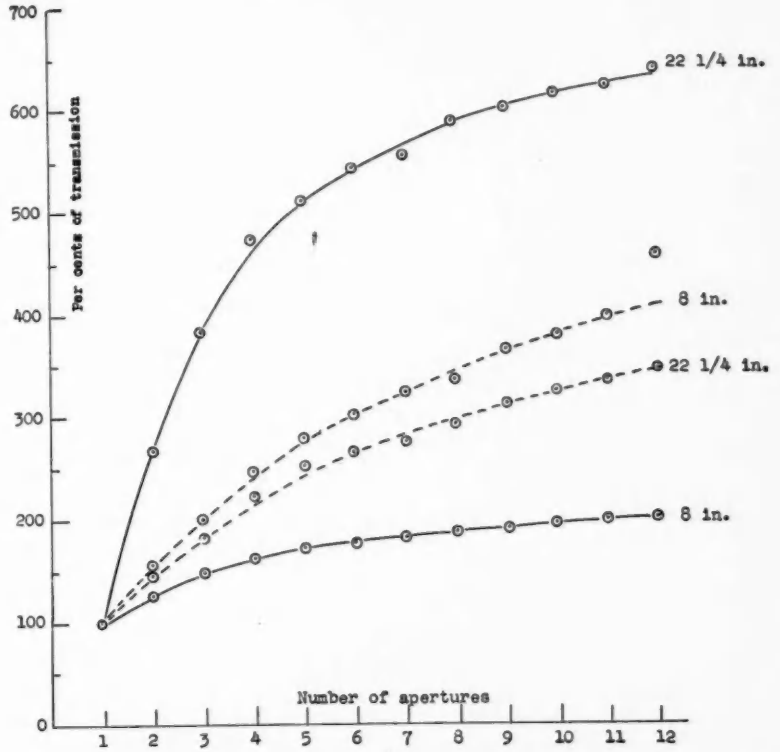

FIG. 4. *Transmission curves relative to one aperture showing the effect of microphone distances.*
——— *UT_4 pipe curves*
- - - - - *UT_3 pipe curves*

The per cents of transmission used in plotting Figure 4 were calculated with respect to one aperture as one hundred per cent. Thus all curves have a common point at one hundred per cent, a fact to be noted when studying the results. If curves were plotted between the actual transmissions and the number of apertures, they could not be compared, for in some cases they cross each other, and also the various series of values have no related points for comparison. Using this basis elimin-

ates any fluctuations that may occur in the amplifier and its relative output, thus permitting direct comparison between curves. The numbers at each curve give the respective distances in inches of the microphone behind the apertures.

The curves in Figures 3 and 4 are of the same form and can be fitted by a cubic equation of the form

$$y = a_0 + a_1x + a_2x^2 + a_3x^3.$$

The constants were each evaluated and then put directly in the form equation. The one-fourth inch curve in Figure 3 was fitted with the following results,

$$y = .06439x^3 - 1.74x^2 + 17.09x - 4.93.$$

Similarly the one-half inch curve in Figure 3 was fitted with the following results,

$$y = 66.947 + 39.89x - 4.1702x^2 + .15447x^3.$$

The constants depend upon the units used and the sizes of the apertures.

Figure 4 shows a greater transmission for the UT_4 frequency when the microphone is at a distance of twenty-two and one-fourth inches than when it was at eight, or eleven and one-half inches from the plate. The same figure shows this condition to be just reversed when a source of UT_3 frequency was used. In other words we have here the greater response for closer distances, which is in agreement with the inverse square law for free space. The UT_4 frequency responses indicate the existence of some interference phenomena. The data obtained when the microphone was at a distance of twenty-two and one-fourth inches from the plate shows results of reenforcement, or greatly increased intensity in the sound waves affecting the microphone. These results were probably due to several causes. A discussion of the causes of interference follows. The apertures used were so small compared with the wave lengths of the sound waves, that the receiving room was within Arago's White Spot resulting from the aperture as a transmitter of waves. The distribution of sound waves offers nothing as a cause of interference.

Foley[1] has shown that the velocity of sound in pipes of small diameters is retarded more than in those of larger diameters. The velocity is less through tubes such as were used in this work, than it is through

[1] Foley, A. L., Proceedings of Indiana Academy of Science, 1919. "Velocity of Sound Waves in Tubes."

free air. On the other hand the velocity of sound through steel is much greater than through open air space. Thus the combined effect of the transmissions through the apertures and the steel plates may be waves of smaller or larger amplitudes depending upon whether the waves from the two new wave sources happen to be in or out of phase in the receiving room. This explanation is not adequate to account for the interference conditions, especially for thin plates and short apertures. However, for longer apertures and thicker plates the effect might become noticeable.

Another explanation of the cause of interference might be that the plate acts as a vibrating body and has a natural period which in turn may be slightly affected by the impetus given by the oncoming waves. The outgoing waves from the plate probably have a different wave length as compared with those going through the apertures so that there would be two different frequency waves in the receiving room. The combination of these waves would give points of reenforcement separated by regions of diminished intensity. Data was obtained to show that changing the mass of the plate and keeping all other conditions invariable does affect the responses from the microphone. This bit of data is a justification for this explanation of the causes of interference. In either this or the former explanation the responses might vary as the as the microphone distances from the plate were made different. A cause of the changing results when the microphone is moved farther away from the apertures is that it subtends a smaller area of the spherical waves from the aperture but is in turn including a greater effective area of the waves going from the plate. The latter explanation is the more probable one.

A third possible cause of phase differences between the waves through the apertures and those from the plate might be in a surface effect analogous to the diffraction of light in passing from one medium into another. There is no data to verify this explanation.

The curve marked eight inches for UT_4 frequency shown in Figure 4 is an example of the results of partial destructive interference.

The results for the transmission of sound through an increasing number of apertures shows that one aperture was most effective when existing alone. As the number of apertures increased the effect of the added aperture gradually lessened. The curves plotted from results appear to become asymptotic as the number of apertures are increased. The cubic equation which was fitted to the curves makes possible the immediate calculation of per cents of transmission when the constants

under given conditions were known and the number or apertures specified.

III. TRANSMISSION THROUGH VARIOUS SIZED APERTURES

The effect of increasing the diameter of an aperture was investigated by using steel plates of the partition window size through which the apertures were cut. The different thicknesses of the plates were one-sixteenth, one-eighth, and one-fourth inches. To get still greater lengths of apertures and keep the infinite plate boundary conditions, the one-fourth inch plate apertures were backed by brass tubing giving total lengths of three-fourths, six, and twelve inches. The tubing diameters were three-sixteenths, one-fourth, five-sixteenths, three-eighths, seven sixteenths, and one-half inches.

The diameters of the apertures used were increased one-sixteenth of an inch each time until a one-half inch aperture was reached. Similar measurements were made for the different plates of various thicknesses.

The plates containing the apertures were largely covered by balsam wool sheets on the faces with only a small open area adjacent the aperture. For the six and twelve inch length tubes interference phenomena were so troublesome that still other covering materials were used to minimize the plate transmissions. Sheets of sponge rubber two inches in thickness were placed over each face of the one-fourth inch steel plate.

Constant frequency sources of 400, 900, 950, 1400, and 1500 v.p.s. were placed at a distance of forty-eight inches in front of the apertures while the swinging pipe sources of UT_4, FA, and UT_3 were placed at a distance of seventy-two inches. The microphone was placed one-half inch behind the apertures in order that the greatest effect of the aperture transmission could be measured, and at the same time get the least effect from the plate transmission. Since the transmission results of the of the apertures were to be compared with open window readings the microphone had to be in approximately the same position as was the aperture.

Figure 5 shows some curves obtained by plotting per cents of transmission against the square of the aperture diameters. An observation of the curves shows a linear relationship to be existing in almost every case that could be fitted by the equation $y=b+ax$. The constant "a" became a negative quantity with increasing destructive interference as was the case for 1500 v.p.s. curve. The constant "b" became negative in some instances because of greater destructive interference conditions.

In several cases no doubt there was constructive interference. Such causes of interference under similar conditions were explained in part two of this paper. The value of the constant "*a*" under these conditions

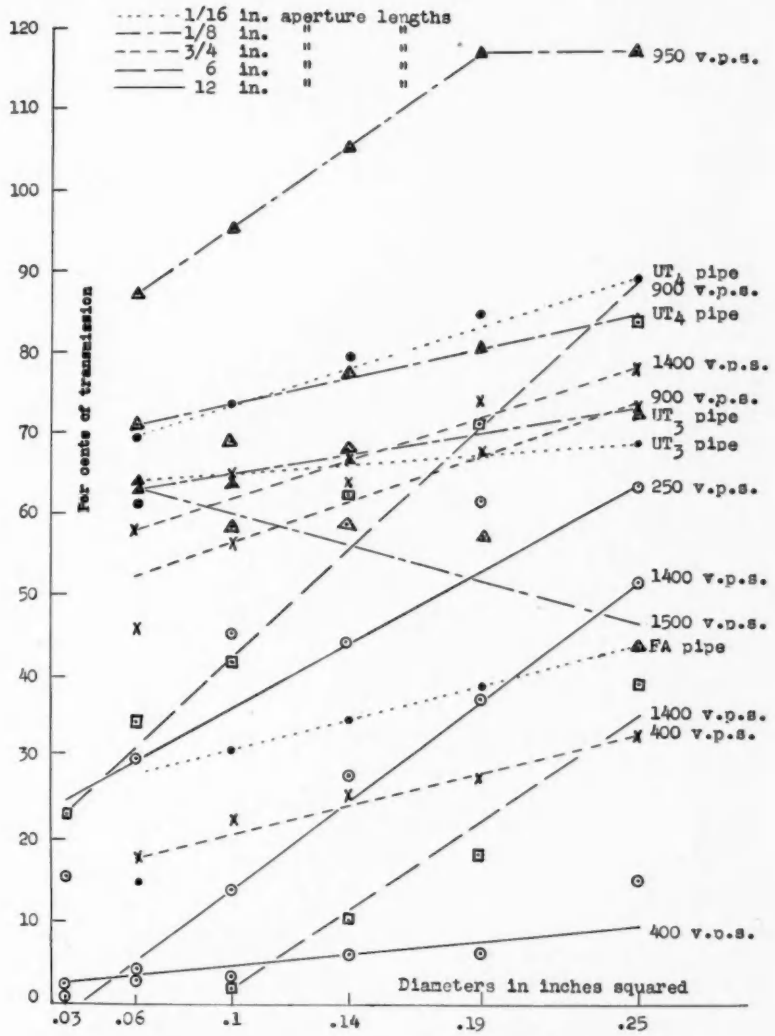

FIG. 5. *Transmission curves relative to open window values showing the effects of the size of apertures.*

was simply a proportionately larger positive amount. An evidence of such phenomena was shown by the slope of the curves. An example of this phenomena may be curve 900 for six inch tubes.

An observation of data bears out the validity of the exponential re-

lationship discussed in Part I of this paper, as long as the length of the aperture does not exceed the diameter. The per cents do not agree since the proper open window evaluation could not be obtained.

CONCLUSIONS

(1) The prediction of Lamb concerning the fraction of sound transmitted by an aperture in an infinitesimally thin plate is in agreement with experimental results. (2) The effect of increasing the length of the aperture, or in other words, the thickness of the plate, results in an exponential relationship existing between the various plate thicknesses and the per cents of transmission. The equation expressing this relationship is $y=be^{ax}$. The per cent of transmission for any specified thickness of plate within the range of this study can be immediately calculated by the equation $x=73.308-7.4084 \log y$, where "y" is plate thickness in inches and "x" is per cent of transmission. These inferences pertain to apertures whose diameters exceed their lengths or the plate thicknesses.

(3) The effect upon the transmission of increasing the number of apertures in a plate was found to obey a definite relationship between the per cents of transmission and the number of apertures. The equation for expressing this relationship was a cubic equation of the form $y=a_0+a_1x^2+a_2x^2+a_3x^3$. The constants vary for various sizes of apertures. For one-fourth inch apertures the equation was $y=.06439\ x^3-1.74\ x^2+17.09\times-4.93$. For one-half inch apertures the equation was $y=66.947+39.89\ x-4.1702\ x^2+.15447\ x^3$.

(4) The transmission for apertures of various sizes and lengths was found to be directly proportional to the area of the apertures.

THE DETERMINATION OF ABSORPTION COEFFICIENTS FOR FREQUENCIES UP TO 8000 CYCLES

By F. L. Hopper

Electrical Research Products, Inc., Los Angeles, California

The effect of certain materials upon the characteristics of sound was recognized as long ago as 1843 when Dickens wrote in "Martin Chuzzlewitt" a description of the location in which Todger's Boarding House was situated as follows: "Mansions now only used for store-houses were dark and dull, and, being filled with wool and cotton and the like, such heavy merchandise as stifles sound and stops the throat of an echo, had an air of a palpable deadness about them." Sabine's early work consisted in a large part in classifying and determining the acoustic properties of materials. Since his time the development of electro-acoustic apparatus and an improvement in experimental technique has greatly increased our knowledge of the properties of these materials. This paper is concerned with a still further extension in this field, and is primarily concerned with the determination of absorption coefficients for frequencies up to 8000 cycles by a chamber reverberation method. With the diminution of the factor of novelty in sound pictures and broadcasting, and an increasing demand for intelligibility and naturalness, more and more time has been spent in advancing the interest in these arts through the extension of the range of recorded and reproduced frequencies. Naturalness and intelligibility are increased by the addition of frequencies above 5000 cycles.* Consequently with the development of apparatus capable of handling a wider frequency band, a knowledge of acoustical conditions over an equally wide band of frequencies is required. The necessity for the determination of absorption coefficients for higher frequencies follows naturally from these factors. Measurements for the range of frequencies below 128 cycles are also of interest and are made whenever materials are encountered which have an appreciable amount of absorption at these low fequencies. If materials are extremely low in absorption at these frequencies it has been found difficult to secure satisfactory measurements due to the very irregular way in which the sound intensity decay occurs.

It has been found that the absorption of sound in air is an important factor in the determination of coefficients for the higher frequencies by

* "Audible Frequency Ranges of Music, Speech and Noise"—W. B. Snow, Bell Telephone Laboratories, Journal of the Acoustical Society of America—July 1931.

the chamber reverberation method.* This absorption is not only a function of frequency but of humidity as well. Since the variation with humidity becomes of increasing importance as the frequency increases all measurements above 1000 cycles made in the past by the chamber method, which neglected this factor, may be seriously in error. A careful technique of measurment must be adopted to secure an accurate determination of coefficients for frequencies above 1000 cycles.

Absorption coefficients are usually determined in three ways: by chamber reverberation measurements, by field measurements in enclosures before and after the installation of acoustic materials, and by the tube method. Probably the most satisfactory method at the present time is the chamber reverberation method, since relatively large test areas may be used giving a more representative sample, and since they may be measured under the same conditions as those under which they are to be used. If the materials are to be used in a wall structure in such a way that panel vibration plays a part in the determination of the absorption, such a wall structure may be built in the reverberation chamber. Usually the reverberation chamber is well insulated, consequently measurements may be made over a wide range of sound intensities, thus increasing the accuracy of the measurements. Measuring equipment may be used such that the coefficients can be determined for a wide range of frequencies. Field measurements may be subject to interference from extraneous noises. The tube method** while useful for comparative tests, is necessarily restricted to small samples. The coefficients obtained by this method in general are not in agreement with the coefficients determined by the first two methods. This may be due in some degree to the fact that the material in the tube is rigidly backed and that the coefficient is determined only for sound waves falling normally upon the material.

The floor plan of the Hollywood Acoustical Laboratory of Electrical Research Products, Inc. is shown in Figure 1. The building will be seen to be comprised of three rooms, a reverberation chamber, a smaller room adjoining it, used for transmission measurements, which may be shut off from the reverberation chamber by means of steel doors, and another room containing measuring equipment.

* Ein Neus Automatisches Verfahren der Nachhallmessing, Zeits. f. Technische Physik, 7, 253, 1930. Meyer.—also—"The Effect of Humidity Upon the Absorption of Sound in a Room, and a Determination of the Coefficients of Absorption of Sound in Air"—V. O. Knudsen—Journal of the Acoustical Society of America—July 1931.

** Wente: "Measurement of Acoustic Impedance and Absorption Coefficients"—Bell System Technical Journal—January 1928.

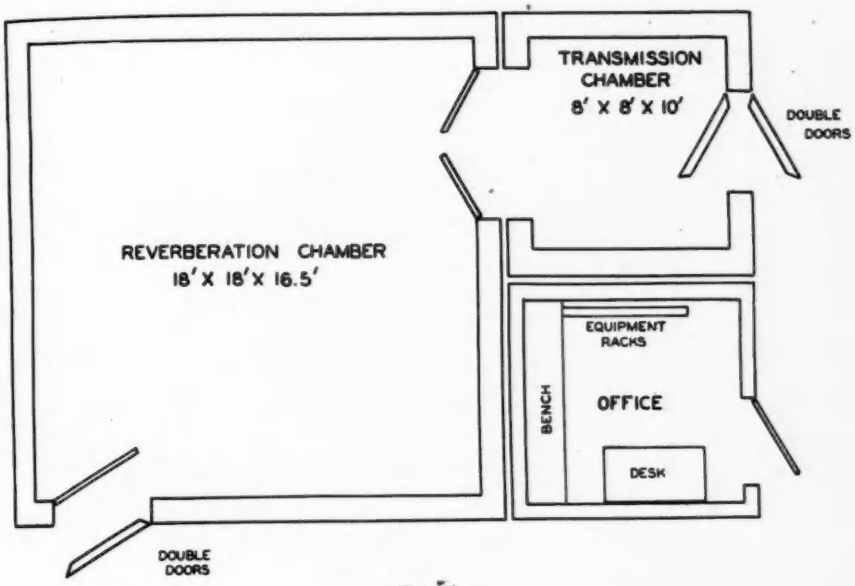

FIG. 1.

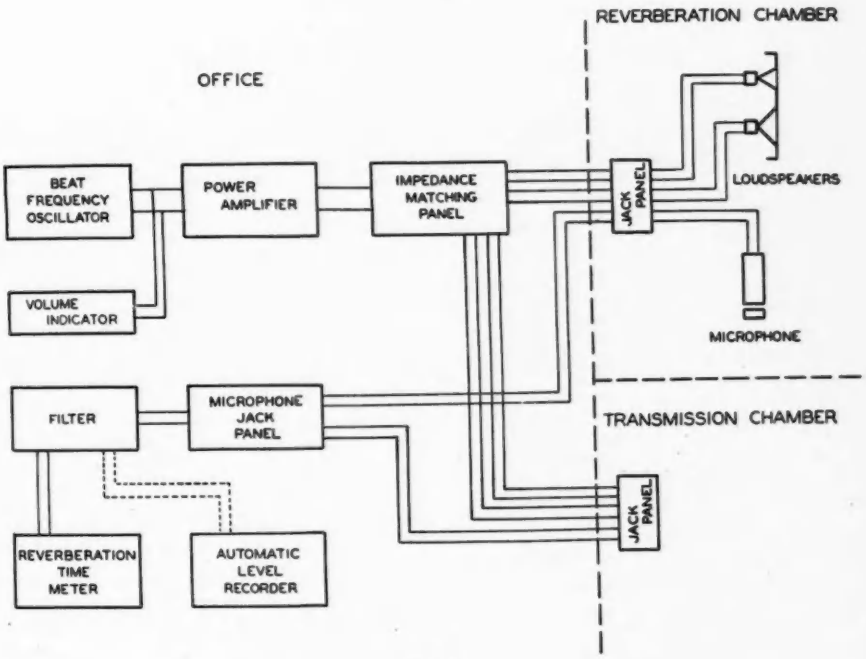

FIG. 2.

The reverberation chamber is a single walled reinforced concrete building 18′ square, having a ceiling height of $16\frac{1}{2}'$, the walls being 12″

thick. The interior of the chamber has a hard smooth finish which contributes materially towards securing long reverberation times. The room used for transmission tests is connected to the reverberation chamber by means of an opening 6′ square which may be closed by steel doors. Materials to be tested for sound transmission are placed in this opening. The transmission room is accoustically treated so that the sound transmission through the panel under measurement is essentially

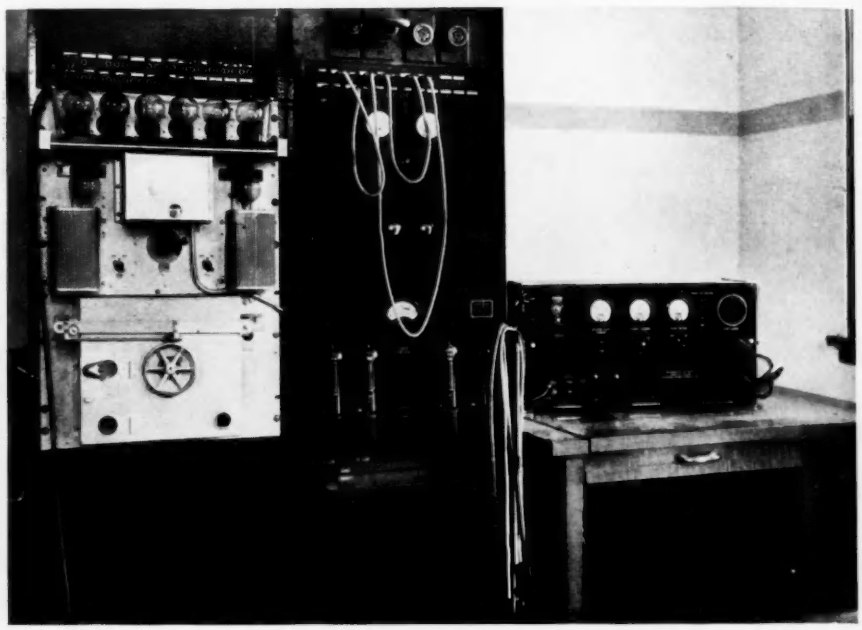

FIG. 3.

unidirectional. Both the reverberation chamber and the transmission room are constructed upon separate foundations and are not structurally connected.

The office adjoining the reverberation chamber contains the measuring equipment, a schematic showing the equipment layout is shown in Figure 2. The same source of sound is used for both reverberation and transmission tests. It consists of a beat frequency oscillator capable of providing frequencies varing from 20 to 9500 cycles. The oscillator is arranged so that a warbling frequency tone may be produced, this being effective in minimizing the effect of standing waves. The warble frequency band width is about 50 cycles, while the frequency of modulation is approximately 400 RPM. The output of this oscillator is further am-

plified by means of a power amplifier which supplies energy to the loudspeakers.

Two loudspeakers are used; one a dynamic moving coil type for frequencies up to 2000 cycles; and the other a special high frequency loudspeaker* which is used for frequencies from 2000 to 8000 cycles. Both loudspeakers are mounted in a box which serves as a baffle for the low frequency speaker.

The measuring equipment consists of a condenser transmitter and its associated amplifier and an electrical reverberation time meter, (a description of this meter has been given elsewhere).** Electrical and acoustical filters are sometimes used in conjunction with the meter and microphone and serve to exclude extraneous noises which might possibly pass through the walls or floor of the reverberation chamber.

A recording level indicator is provided for use in transmission measurements and may be connected to the microphones in either the transmission or reverberation rooms. Figures 3 and 4 are photographs showing the measuring equipment and the interior of the reververation chamber.

The absorption coefficient for any material is determined by first measuring the reverberation time in the empty chamber for all the frequencies for which coefficients are to be found. The reverberation times are again measured when a definite amount of surface in the chamber is covered by the material under test, the material usually being placed on the floor or on one of the wall surfaces. The absorption coefficient may then be computed from these measurements as follows: let α be the average coefficient of absorption for the empty chanber; S the total surface for the empty chamber; α_1 the average coefficient of absorption for the chamber when an area S_1 is covered by an absorbing material, α_2 the average coefficient for the material under test, then:

$$\alpha_1 S = \alpha S + \alpha_2 S_1 - \alpha S_1$$

hence:

$$\alpha_2 = \frac{S(\alpha_1 - \alpha)}{S_1} + \alpha,$$

i.e. the absorption coefficient for the sample under test is equal to the

* Bostwick: "An Efficient Loudspeaker for the Higher Audible Frequencies"—Journal of the Acoustical Society of America, October 1930.

** "Measurement of Reverberation Time and Its Application to Acoustic Problems in Sound Pictures"—F. L. Hopper, Journal of the Acoustical Society of America—April 1931.

total number of units in the chamber with the material present minus the total number of units for the chamber empty divided by the area

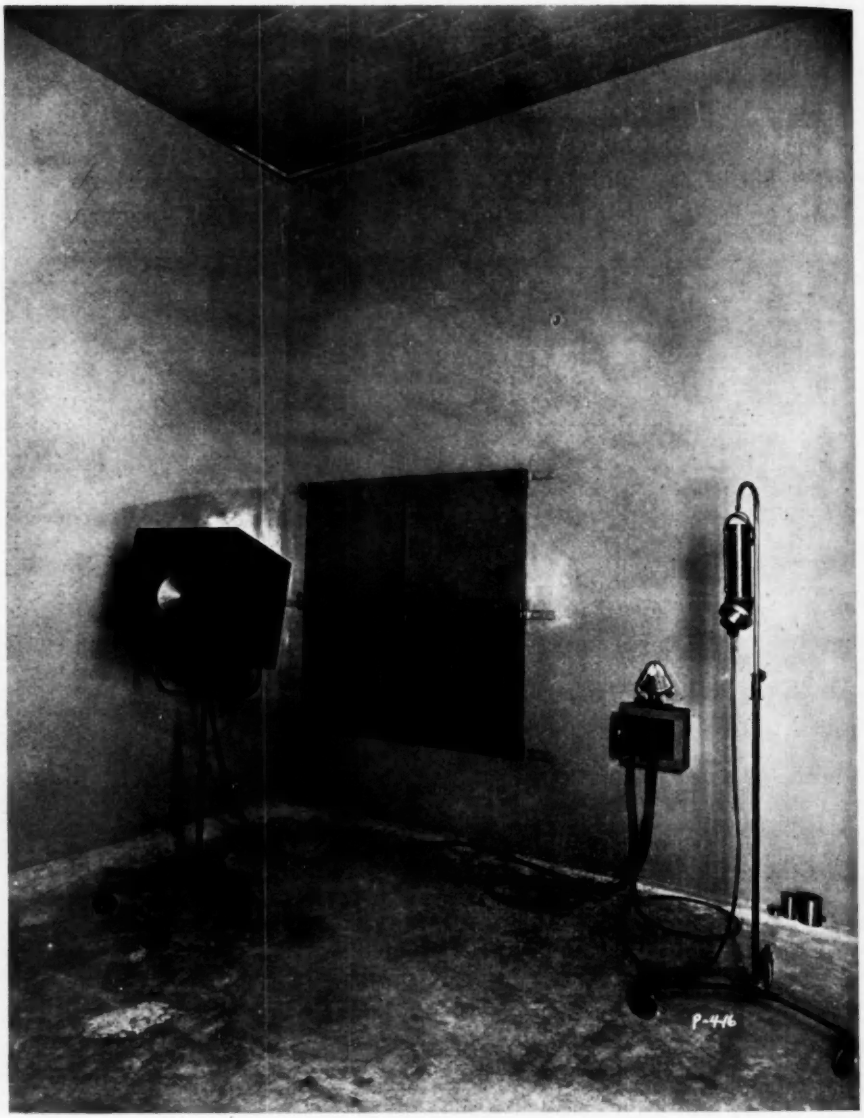

FIG. 4.

of the test sample, plus the absorption coefficient per square foot for the chamber empty. α and α_1 are easily determined by means of Eyring's Reverberation Formula:

$$t = \frac{.05}{-S \log_e (1 - \alpha_a)} \cdot \frac{v}{}$$

Where t is either the reverberation time measured in the empty chamber or the time for the chamber with the material present. In all of the reverberation time measurements the times are determined from the average slope of the sound intensity decay curve, the curve being plotted in small successive steps over a wide range of sound intensites. Two

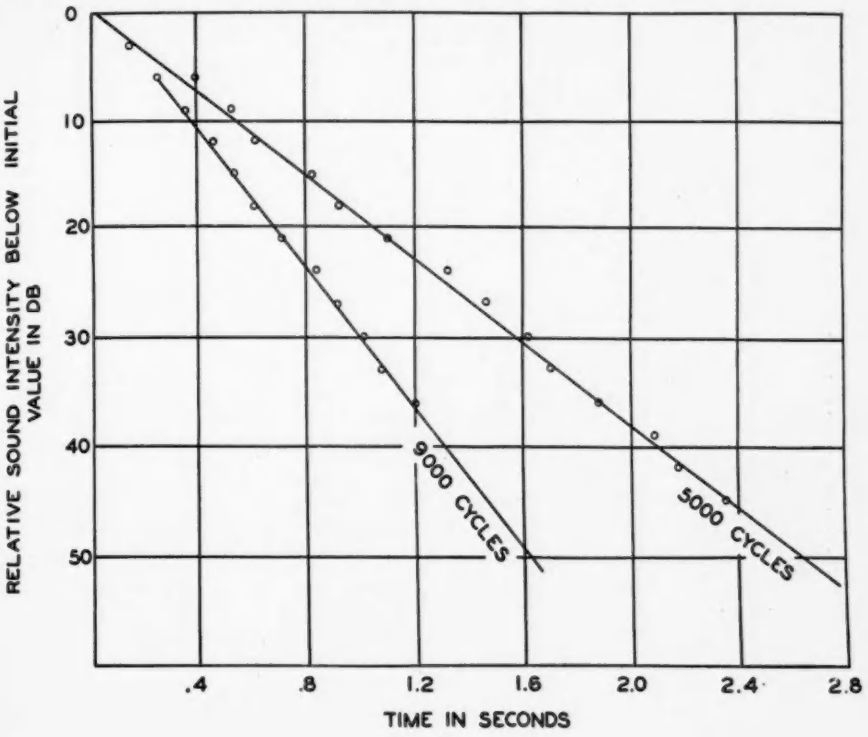

FIG. 5.

typical sound intensity decay curves are shown in Figure 5. The points closely follow a straight line indicating that the sound in the chamber is rather thoroughly mixed. Measurements are usually made for several positions of the microphone in the chamber in order to insure a good average value being obtained for the measured reverberation time. Usually the variation in time with microphone position is negligibly small for measurements at frequencies higher than 1000 cycles.

Past practice has in some cases been to determine the reverberation time for the empty chamber from the average of a number of tests and subsequently assumed to remain constant. While this may be true for

the lower frequencies it is certainly not true for the higher frequencies where sound absorption in air plays a large part in determining the total amount of absorption in the chamber. The magnitude of this effect depends both upon frequency and humidity. It is necessary, therefore, to calibrate the room empty each time a measurement is made or to provide an elaborate calibration for the empty chamber, which shows the total amount of absorption in the chamber as a function of humidity and frequency. If this is not done serious errors may occur for measurements at frequencies as low as 1000 cycles. An investigation of the variation of

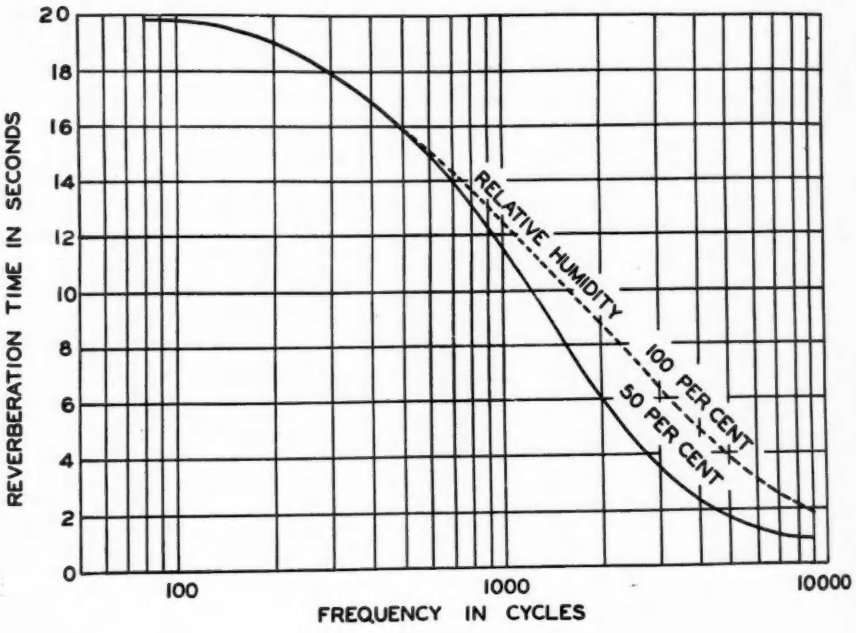

FIG. 6.

absorption with humidity was begun in January of 1931. An example of the variation in reverberation time with frequency for two conditions of humidity is shown in Figure 6. It will be noticed that there is a slight difference in reverberation times at 1000 cycles for the two conditions of humidity, this difference in time corresponding to a change in absorption of two units. At 3000 cycles the difference in times is equivalent to a change in absorption of 31 units, and at 9000 cycles is 126 units. Since the variation in the total absorption in the chamber, due to changes in humidity alone, may be much greater than the total absorption introduced by the material itself, the necessity for taking this factor into consideration can readily be appreciated.

Observers have noticed in the past that the apparent absorption coefficient for a given material increases as the area of the test sample is decreased. Inasmuch as the coefficients so obtained may be in excess of unity, it is the practice of this laboratory to use fairly large test samples. This also offers the advantage of securing a more average sample of the product, and gives a larger change in reverberation time. If materials are low in absorbing efficiency the use of large samples will increase the accuracy of the determination of the coefficients. With the development of electrical reverberation time meters no obstacles are encountered

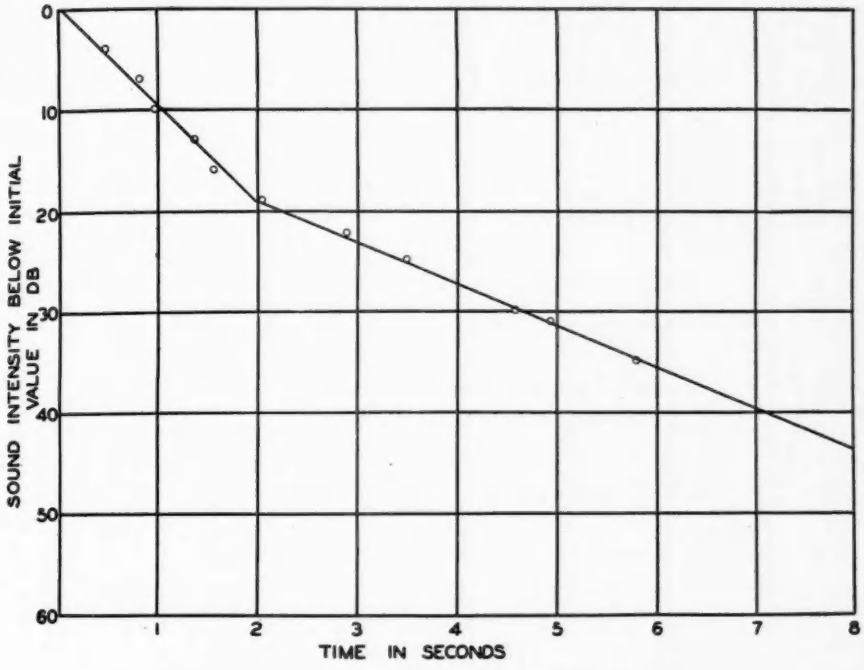

FIG. 7.

in the measurement of the shorter reverberation times which may be obtained when using such samples. If the materials are highly absorbing it is essential that the sound intensity decay curve be plotted in small successive steps in order that multiple rates of decay may be detected should they exist. When one surface of the reverberation chamber is highly damped such a condition may occur. The measurement of only a single drop in intensity would obviously lead to an incorrect reverberation time in such a case. An example of two distinct rates of decay obtained when measuring a material are shwon in Figure 7.

Coefficients of a given material determined in this laboratory upon

successive occasions are within 5 per cent of any individual measurement. It has been found that certain types of compressible materials, such as carpets, vary in absorbing power depending upon the amount of compression they have undergone. By passing a roller over the nap of such a carpet the coefficient was reduced 15 per cent over the unrolled state. Shaking it restored the original coefficient. Porous, flexible materials placed upon furring strips do not, in general, have the same coefficient as when rigidly backed. This applies equally well to rigid materials mounted in such a way that panel vibration plays a part in determining the total absorption. Materials which are hygroscopic may vary in absorption depending upon the amount of moisture which they contain. The exact magnitude of this effect might be difficult to consider in commercal measurements due to variety of factors involved. It is believed that the variations in coefficients determined in various laboratories may be due to such factors as these. Summarizing, possible differences may be found in the following:

(a) Failure to calibrate room empty for each measurement, i.e. failure to consider variations in absorption in the chamber due to humidity.

(b) Methods of mounting materials under test.

(c) Variations in absorption due to moisture content of materials which are hygroscopic.

(d) Variations in absorption due to a change in physical state such as may be caused in soft materials by compression.

(e) Variations in absorption due to a variation in the area and arrangement of the test samples used for measurement.

If laboratories would state definitely the exact conditions under which materials are tested it should greatly facilitate any comparison of data which it might be desirable to make.

The coefficients for a variety of acoustic materials have been measured for frequencies up to 8000 cycles, the first commercial measurement having been made by Electrical Research Products in February 1931. From the test data to date it is not possible to tell whether or not there is any definite relation between the absorption at the lower frequencies and that obtained at the higher frequencies. Consequently, it is difficult to predict just what may occur for the higher frequencies from a knowledge of the coefficients for frequencies below 4000 cycles. Some materials show an increase in absorption from 4000 cycles to 8000 cycles while others exhibit a marked decrease in absorption. The absorption frequency characteristics of several types of commercial materials are shown in Figures 8, 9, and 10.

Figure 8 illustrates two types of materials; one a precast plaster tile, the other a rock wool fill placed between 2″×4″ studding and secured

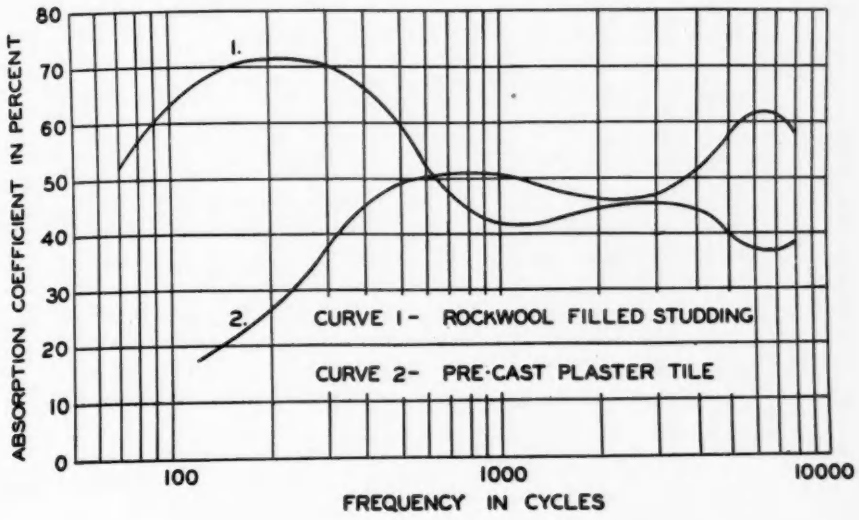

FIG. 8.

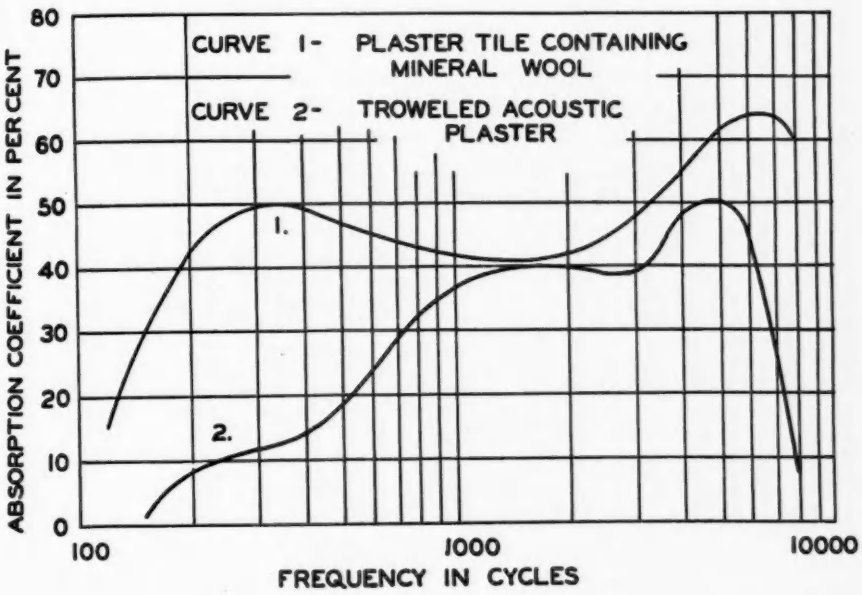

FIG. 9.

in place by muslin. The plaster tile shows an increase in absorption with frequency while the rock wool structure has its maximum absorp-

tion for the low frequencies and decreases in absorption considerably for the higher frequencies.

Figure 9 shows the characteristics of a troweled on acoustic plaster, and an acoustic tile containing mineral wool. The troweled plaster shows a decreasing amount of absorption for the high frequencies, while the acoustic tile is fairly uniform in absorption.

Figure 10 shows the absorption for a heavy carpet and a hairfelt pad. Both of these materials have high absorption at the high frequencies. MacNair* has prepared an ideal absorption frequency curve for acoustic

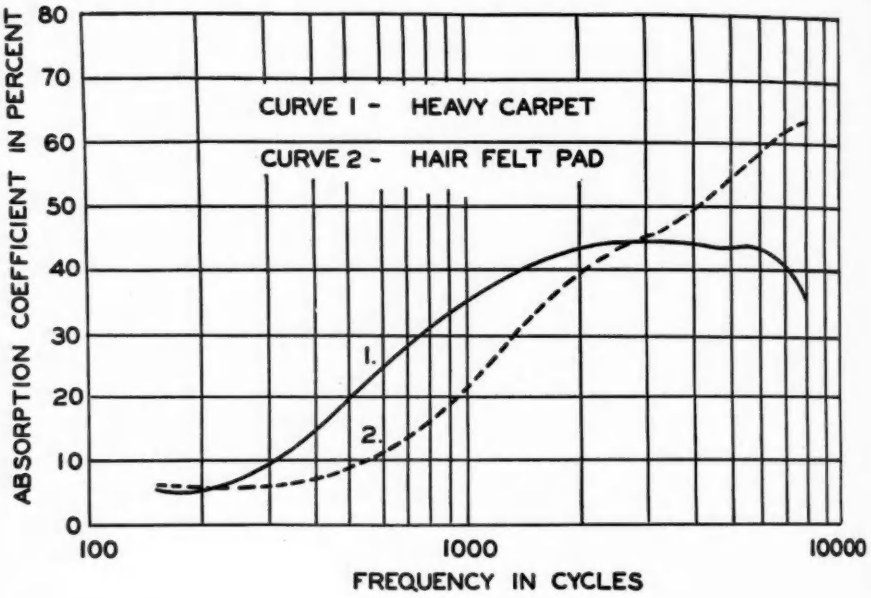

FIG. 10.

materials based upon loudness. This curve represents the relative number of absorption units which a material should have for frequencies varying from 50 to 8000 cycles. It is assumed that this material provides the major portion of the total absorption present. If one the other hand a given material provides only a portion of the total absorption, it will be the summation of the absorptions of all the contributing materials which should approach this ideal characteristic. If the materials shown in Figures 8, 9 and 10 were used to supply the greater part of the acoustic treatment in some particular case then we may compare their characteristics with this ideal, and note the variation.

* "Optimum Reverberation Time for Auditoriums"—Journal of the Acoustical Society of America, January 1930.

Consideration of all of the various factors which affect the determination of absorption coefficients at the higher frequencies would make it appear necessary to carefully review some of the data which has been obtained in the past, relating to these same coefficients.

Materials having various types of frequency absorption characteristics are necessary for such different classes of work, as: the reduction of noise, the acoustic treatment of theatres and auditoria, and the treatment of rooms and stages used for the electrical recording of speech and music. In order to intelligently design acoustic treatment for these classes of work it is essential that this information be available.

THE TRANSMISSION OF SOUND THROUGH SEA WATER*

By Herbert Grove Dorsey

Principal Electrical Engineer, U. S. Coast and Geodetic Survey

The manner in which subaqueous sound travels from one point to another and the velocity of such travel has become a subject of vital interest to members of the Coast and Geodetic Survey, as the methods of surveying depend to a greater extent upon this branch of acoustics. This is especially true in Radio Acoustic Position Finding,[1] a method of determining the survey ship's position in offshore hydrography.

In obtaining data for making nautical charts, it is necessary not only that the depth of the water be measured, but also that the location of the depth measurement be determined. When surveying near the shore, the position of the vessel is determined by a graphic solution of the three-point problem; that is, two sextant angles, right and left, are simultaneously obtained by two observers sighting on three objects, the positions of which are accurately known from previous land surveys. One observer measures the angle subtended by the center object and that to the right while the other observes the angle subtended by the center and left objects, each observer using the center object in common. These angles are then plotted on a "boat sheet" in the pilot house or chart room. For this purpose a large sheet of linen-backed paper is used on which lines have been carefully ruled corresponding to the polyconic projection of latitude and longitude of the particular region to be surveyed. This projection also has plotted on it the exact location of the shore line and all objects, or signals as they are called, which are to be used for observational points during the survey of that particular area. On the Atlantic Coast of the United States, most of these signals are built in advance of the hydrographic surveying, some as high as 120 feet. Whenever possible, lighthouses, towers, smoke stacks or other prominent high objects are used as signals, provided their positions have been accurately determined in the general control triangulation. These signals can seldom be seen more than 12 or 14 miles on the Atlantic Coast where the shore has little elevation, while on the Pacific Coast and in the Philippines, mountain peaks may be available for signals visible for 75 miles or more.

Immediately upon being observed, the right and left angles are set

* Publication Approved by the Director of the Coast and Geodetic Survey of the U. S. Department of Commerce.

[1] Special Publication No. 146, Government Printing Office 1928.

on the right and left arms of a three arm protractor, which is then moved about on the boat sheet until the three arms touch the respective points representing the three signals. A needle point at the center of the protractor is then depressed, making on the boat sheet a tiny dot, which represents the position occupied by the ship when the angles were observed.

Meanwhile the ship has been moving forward at a speed of ten to fourteen knots and at regularly timed intervals depth soundings have been taken by the Fathometer,[2,3] when the water is deeper than 15 fathoms, or by hand lead if shallower, in which case the ship's speed must be decreased to about 5 knots. Positions are thus determined at fixed periods of five or ten minutes, or at such intervals as may be necessary to maintain the ship's position on the sounding line, changing course whenever it is found by successive plotted positions that wind or currents make a change necessary. Between positions thus determined, depth measurements are plotted on the sheet, spacing the distances between soundings to agree with the speed of the ship between positions. Depth curves or submarine contours are also drawn on the boat sheet to determine whether any irregularities exist, such as shoals or valleys. If any are indicated, extra lines or system of lines of soundings are run over the area so as to thoroughly develop the irregularities while the vessel is in the immediate vicinity.

The ship runs a regular system of parallel sounding lines over the area to be surveyed, spaced at varying distances apart, depending upon the character of the bottom and the depth. In an important, confined channel in which the depth approaches the draft of vessels using it, or on important shoals on or near sailing courses, these lines are spaced at times as close as 50 meters; whereas, in the open and in greater depths this spacing is increased, even to a mile in depths of about 100 fathoms and in oceanic depths to five miles.

At certain fixed intervals the ship is brought to a dead stop and the depth measured with a lead weight on a piano wire about 0.64 mm. diameter. At the same time a sample of the bottom is obtained, water temperatures measured at surface, mid-depth and bottom, and salinity determined for the necessary correction to the depths as determined by the fathometer.

In this optical method of obtaining the ship's position, it is evident that visibility must be sufficient to see the shore signals, which pre-

[2] J.O.S.A. & R.S.I. Vol. 9, Sept., 1924.

[3] U. S. Patent No. 1, 667, 540, 24 April 1928, by the author.

cludes any work during fog or at night, or, in any case, where distances are so great that the earth's curvature prevents direct sight. Prior to 1924, the ship's positions for surveying at a considerable distance from shore were determined by a method known as precise dead reckoning. The sounding line was started from an inshore position run offshore by course and distance, allowance being made for currents and leeway, and back to another inshore position. The closing error; that is, the difference between the end of the line as determined by the dead reckoning and as determined by the absolute position, was distributed proportionately back through the line of sounding. This method of precise dead reckoning, while highly developed, can be used only in good weather and at best the vessel's position is subject to the vagaries of weather and currents.

As a result of work started during the war, the Radio Acoustic Position Finding method was developed. In determining positions by this method two hydrophones are placed in the water from 10 to 50 nautical miles apart and their positions determined. They may be near the shore, in which case each hydrophone is supported above a concrete block resting on the bottom and connected to a temporary shore radio station by a cable, or each hydrophone may be suspended in the water from a float near a station ship, anchored at a known position. In either case the hydrophone is connected to an audio frequency amplifier which will automatically actuate a radio transmitter by a relay or a thyratron whenever a sound of sufficient intensity reaches the hydrophone.

When it is desired to determine the mobile survey ship's position, a bomb with lighted fuse is dropped overboard from the vessel, the bomb varying in size from a blasting cap only, when the distance is small, to a half pint can, pint or quart can of TNT or even a cast iron bomb of something over a quart of TNT when the distance to the hydrophone station is great. At the time the bomb is dropped overboard, a two pen chronograph tape is started on the mobile surveying ship. One pen marks seconds of time from a chronometer and the other marks the instant that the sound of the explosion reaches a hydrophone on the survey ship. The sound proceeds to the two hydrophone stations, causing a radio signal to be emitted from each, and both signals are received at the survey ship and recorded on the chronograph tape. The elapsed time intervals are measured to hundredths of a second and, having experimentally determined the velocity of sound in water in that vicinity, the distances are readily computed and the arcs swung on the boat sheet with a beam compass, the intersection of the two arcs as deter-

mined by the distance from each hydrophone giving the location of the ship's position at the time when the bomb was dropped overboard. Positions are thus determined at fixed intervals the same as by the visual method and the line plotted with the depths on the boat sheet as previously described.

By the radio acoustic method of determining positions surveying can proceed in any condition of visibility, day or night, sun, fog or rain. The floating hydrophone stations can be placed at any position in which they can be anchored, even hundreds of miles from shore, as is now being done in the survey of Georges Bank. On this survey a primary buoy is planted and its position determined by the weighted average of a considerable number of astronomical observations, and then a regular system of triangulation, with sides 10 to 15 miles in length and with a buoy at each vertex is extended along the Bank from this primary buoy, the locations of the buoys being determined by bombing the distances between then. The hydrophone station ships then anchor at any pair of buoys, the relative positions of which furnish the strongest control of the surveying ship. It might be mentioned in passing that the ever changing water currents on Georges Bank introduce many difficulties not encountered in ordinary surveying.

The plotted boat sheet, as made according to the foregoing brief description serves as a guide for the marine surveyor in determining whether the area has been satisfactorily and uniformly covered by the system of sounding lines and constitutes what might be called a rough outline of the surveyed area. From the recorded data, a "smooth sheet" is now prepared on which all positions are more accurately plotted and all corrections to depth measurements applied, including density and temperature of the sea water and corrections for height of tide at the time of measurements. These sheets and all pertinent records are then forwarded to the Washington office of the Coast Survey where they are carefully reviewed and checked in every detail and also with any previous records, every necessary precaution being taken that the data shall be as nearly correct as possible before being placed on a chart for public use.

The general method of Radio Acoustic Position Finding[1] was developed before the author joined the Coast Survey and varying hypotheses had been proposed to explain some of the peculiarities in the passage of sound from the exploding bomb to the hydrophone. In a re-

[1] Loc. Cit.

cent article[4] a hypothesis was advanced that sound from an exploding bomb traveled to the receiving hydrophone by a path of multiple reflections from the sea bottom and the surface of the water.

To the writer this hypothesis appears so different from the way sound travels through air that it seems unnatural. The following theories are offered instead of the reflection hypothesis, based on the writers experience in subaqueous sound experiments during the last 15 years. These theories have appeared in an article in the "Bulletin of the Association of Field Engineers" for December, 1930, an unofficial manuscript publication by the personnel of the Coast and Geodetic Survey for Service distribution.

This theory would have the sound from an exploding bomb, such as is used in Radio Acoustic Position Finding, spread out equally in all directions, just as would occur when a firecracker explodes in mid air. Of course, there is the difference that in water the velocity is greater and the acoustic resistance is less; if the same amounts of sound energy are liberated in air and in water, the effect in the latter medium will travel farther and faster. If any obstruction is encountered, there will be reflection or absorption or both, part being reflected and part absorbed. From the explosion a single compressional wave starts out followed by a single rarefraction wave, having the same sound as an explosion in air; bang, pop, boom, or whatever word describes a single sound before it has suffered reflections or refractions or other distortion. Probably everybody has been near enough to a flash of lightning to hear just one loud report, no mutterings or rumblings such as are heard when farther away. When listening to bombs and detonators on Georges Bank with an electromagnetic receiver and good quality amplifier, at short distances they sound like short quick explosions; when farther away, however, 40 to 60 miles, the sound, of course, is not only much weaker, but also not of short duration but of a prolonged nature.

In warm water sound travels faster than in cold water and the writer is of the opinion that the warm water has greater acoustic resistance so that a given amount of sound energy will be dissipated more quickly and will travel a lesser distance. The first evidence of this appeared to the writer during the winter of 1918–1919, while associated with Dr. John Hayes Hammond, Jr., in the development of dirigible torpedoes. In the clear, cold water at the entrance of Gloucester Harbor, receiving apparatus gave certain deflections at a distance of 5000 yards from the source; but, when the same apparatus was moved to the warm waters

[4] Journal of the Franklin Institute, Dec., 1928.

of Hampton Roads, Va., the same deflections could not be obtained at distances greater than 2000 yards without further refinement and much time was lost in discovering that the difference was due to the water and not to some deterioration of the apparatus.

Experiments in cold Alaskan waters in July, 1927, with improvised apparatus, gave readable deflections at a distance of 40 miles from dynamite bombs and it was then predicted that Radio Acoustic Ranging would be successful; this prediction was amply justified by the satisfactory results which have been obtained there during the past three summers. In the Sulu Sea where the water is warm similar improvised apparatus in December, 1927, received the first bomb of TNT when only a few miles distant, but no sound was heard at any greater distance. Although shoals may have vitiated this test to some extent, the results seemed so similar to many months of trials on the Carolina coast during 1926–27, where it was impossible to get satisfactory results, it is believed that the difficulty in the Sulu Sea was due to warm water.

During the fall of 1929, preliminary tests were made with the Coast Survey Ships LYDONIA and ECHO off the Virginia coast to determine whether a method of Radio Acoustic Ranging could be developed by using floating hydrophone stations instead of fixed stations near the shore which had been found impracticable along the south Altantic coast. These experiments gave promising results when the floating hydrophone stations were located in depths of about 20 fathoms, the sound of bombs being received at distances of 75 miles. It was stated, however, that it might not work so well in the warmer waters along the Florida coast and such proved to be the case in the early spring of 1930, when it proved equally difficult to cover a distance of 20 miles as to get results at 60 miles off the Maryland coast.

During the summer of 1930 in the cool waters on Georges Bank, quart bombs of TNT gave good automatic return signals at 60 miles, the longest distance tried, and many records were made as far as 18 miles with detonators alone. These distances are comparable with those on the Pacific coast, indicating that cold water has less resistance for sound waves than warm water.

During the month of March, 1931, again in the warm water on the east coast of Florida with apparatus similar to that used on Georges Bank, the ships LYDONIA and GILBERT had much better results when the sound of the bombs traveled in a north or south direction than in an east to west direction. Very shortly after the bombing ship traveled eastward into the Gulf Stream, the receiving ship in the cooler

water towards shore could not even hear quarts of TNT. The writer was not present during this survey work and before instructions could be issued to the ships to make tests with both ships in the warm water of the Gulf Stream the season's work was closed.

This effect may have been due to greater absorption in the Gulf Stream or it may have been due to reflection at the junction of the two waters, the boundary at times being very sharp; or the failure may have been due in part to both. At any rate there can be no question of the sudden change in distances attainable as reported by the officers in charge of the work.

The mere fact that the temperature of the water is high may not be the entire reason for its increased resistance. Possibly warm water carries more matter in suspension, such as marine growth, and may have more occluded gases in the form of fine bubbles. It is well known that the water which has been churned up by the propeller of a ship is a regular blanket for sound, absorbing it completely. This is immediately noticeable when watching the fathometer, for as soon as the ship starts to back, so that it enters the disturbed waters, all echoes cease and only ship noises are heard.

Since the velocity in warm water is greater, that part of the sound passing along the upper strata of water will travel faster than that which has passed to the lower strata. It may be visualized as if the sound were traveling like a cloud of dust, filling the space as it goes forward. If a large obstruction be near, such as a battleship, there will be a sound shadow immediately on the other side of it, but farther away the shadow will be less sharp and finally disappear. Such effects were actually observed during the experiments at Hampton Roads when many battleships were there. The same is true in air, except that sound bends around corners more easily then it does in water, so that the sound shadows in air are less sharply defined than in water.

In Radio Acoustic Ranging shoals and capes are encountered more often than battleships. The Commanding Officer of the Coast Survey Ship PIONEER reported, however, that Tillamook Rock made no shadow. This is a massive rock rising in some 30 fathoms of water to a hundred feet above the surface near the coast of Oregon. The ship steamed in the arc of a circle about the shore hydrophone station while sounding its oscillator and the received signals showed no variation due to a shadow effect.

If one imagines a portion of the ocean as a conductor of sound, it is readily seen that, in general, it is a conductor which is broad horizon-

tally and narrow in the vertical direction as would be a rectangular speaking tube. Near the shore, especially on the Atlantic coast, the conductor is wedge shaped, the thin edge being near the shore. Any obstruction such as a shoal will decrease the area of cross-section of the conductor and it will be more difficult for the sound to get through, the shoal acting like a dent in a speaking tube.

If sound starts in a body of water of small cross sectional area at one end and goes towards a greater cross section, the acoustic resistance will be less than when the sound is transmitted from what may be called the large end to the small end. In the first case, the conductor, by spreading out, transmits the sound without absorption by the bottom, while in the converse, more of the sound is absorbed by the inclined surface of the bottom. Experiments between the LYDONIA and ECHO showed that bomb sounds traveled farther from inshore depths of 6 fathoms outwards to 20 fathoms than in the reverse direction.

In spreading out to fill the cross section of any area, that part of the sound which passes through the warm water will travel faster than that portion which travels through cold water, but since the acoustic resistance of the cold water is less the sound energy density will be greater. It is probable that the hydrophone receives a series of pressure changes, the individual parts of which have come through almost the entire cross-section of water. A curve drawn by apparatus of infinite sensitivity would reveal the actual shape, in frequency and amplitude of the received energy.

If the multiple reflection hypothesis is considered for a moment, it might be shown that the velocity and surely the acoustic resistance would be some function of the bottom because different bottoms have decidedly different reflective properties. Experience with the Fathometer[4] shows that the best reflecting bottom is that marked on the charts as soft mud. Apparently this presents a smooth flat compact surface to sound waves forming a nearly perfect reflector. The soft mud bottoms along the Maine coast are much better reflectors than the hard sand bottoms of the North Carolina coast as evidenced by multiple echoes in depth measurements. By multiple echoes is meant that the sound from the Fathometer oscillator goes to the bottom of the ocean from which it is reflected, returning to the flat steel hull of the ship from which it is again reflected to the ocean floor and thence reflected again to the ship. As many as 12 such returns have been heard by the writer, meaning a total of 23 reflections. Only ten returns could be

[4] Loc. Cit.

counted in the Japanese high speed moving pictures of sound waves where a spark was made in an elliptical reflector. No multiple echoes are observed from ships without some portion of the bottom being flat.

It is probable that flat bare rock is a better reflector than mud, but such flat rock bottoms are seldom found. Broken coral is poor and sea grass seems about the poorest, as would be expected. However, with all the different bottoms encountered in Radio Acoustic Ranging, the measured velocities seem to be entirely independent of the nature of the bottom.

Since the velocity is greater in warm water than in cold, and since the warm water is usually on top, considering a line perpendicular to the line of travel, or the wave front, its upper end will be bent forward so as to be headed towards the bottom. This refractive phenomenon is called the temperature gradient effect, and has been so called by the Submarine Signal Company for about 15 years. During that time, it has been their uniform experience that submarine signals are heard farther in winter than in summer, due to the colder water, and also that signals are heard farther in clear water than in muddy water.

In visualizing what takes place at the hydrophone and ensuing apparatus at the RAR receiving station and at the chronograph itself at the recording station, it should be remembered that sound energy of a certain peak value must be received at both stations before a mark is made on the chronograph tape. Taking concrete examples, on the Pacific Coast where a plate circuit relay is used with its amplifier at the hydrophone station, it may require a current change of 2 milliamperes to operate this relay which transmits the radio signal, whereas the RAR receiving stations as now operated on the Atlantic coast require a change of less than 0.1 milliampere to operate a thyratron which transmits the signal. Assuming these to be the minimum values of current change to produce operation, increasingly greater amounts will operate the relay in a shorter time although it will make no change in the time of operation of the thyratron which operates in a few micro seconds, being electronic motion instead of mass motion.

If the distance between bomb and hydrophone is great, the sound upon reaching the latter, having traversed paths of different velocity and of different acoustic resistance, will not be one single impulse like an explosion, but will be a sound of some duration, the first part of which may have less acoustic energy than the last part and it may be the last part only which has sufficient amplitude to operate the transmitter.

The difference in time of travel between top and bottom strata for a distance of 50 miles may, in some cases, be as much as one second.

While in New York City recently, during a thunder storm, two opposite cases were observed; thunder was heard in about one second after a lightning flash, although the maximum intensity was not heard until 11 seconds had elapsed; shortly afterwards another flash preceded its first crash by about one second although lesser rumblings continued for 15 seconds. In the first instance sound shadows probably prevented the maximum amplitude from reaching the ear until the sound had traveled some two miles by reflections, while in the second case there was evidently a direct path for the maximum.

Although no sound shadows probably exist in RAR, the necessary sound pressure to operate the transmitter may be at any part of the duration of the sound from the bomb. It seems evident that the minimum amount of acoustic energy necessary to actuate the transmitter will have arrived via the path of least acoustic resistance for that energy so that the transmitter will be operated by the first portion of the sound which has sufficient intensity. This intensity like the loudest part of the thunder, may be at any portion of the sound duration. This might be designated the first maximum of sufficient sound energy, or primary maximum. This necessarily brings in the measuring apparatus as well as purely theoretical considerations. It may be that in one case the calculated velocity along the bottom will agree with the experimental results, or it may agree with some intermediate water layer. Surely it can never be along the surface because of refraction and of the usual surface disturbances due to sea waves, and also because the bombs are always fired at some depth. In cases that bombs have not sunk but have exploded on the surface, they are seldom heard more than a mile.

The Commanding Officer of the GUIDE has recently reported that, while working near Vancouver Island, from one and only one of three shore RAR stations the measured time of travel of sound from small bombs was greater than that of large bombs. A cast iron bomb of about a quart of TNT gave a measured time interval about a quarter second less than a quart of TNT in a tin container, the latter giving less sound energy than the former. Insufficient data are available to say this may not have been an accidental case, yet unusual conditions of the water could cause it and also the "primary maximum" might be at the end of the sound for the smaller bombs and at the beginnings for the larger ones. With the large bombs the distance arcs from the three shore sta-

tions intersected in a point, as they should, while with the small bombs the arcs formed a triangle of uncertainty.

It may be possible that some of the sound goes into rock strata at the bottom and is carried long distances by rock formation and then reaches the hydrophone. While this may be possible, it is believed improbable in most cases because solid bare rock bottoms are seldom encountered and, if so, they would probably reflect most of the sound instead of transmitting it, and where rock bottoms are covered by a layer of sand the latter would act as an absorbing medium to prevent the sound from reaching the rock.

Since this article was first started the writer has learned that Mr. A.L. Shalowitz, Cartographic Engineer of the Coast and Geodetic Survey, has made an extensive study of the records of measured velocities of several years work in Radio Acoustic Ranging on the west coast and compared them with the calculated velocities of different horizontal layers of water along a vertical plane through the bomb and hydrophone. His studies indicate that both the hypothesis of multiple reflection and of straight line transmission are untenable and the comparisons show that the measured velocities correspond closely with calculated bottom temperature velocities for a large number of independent measurements where the average depth was about 60 fathoms. In other words, from his comparisons it would appear as though the sound went to the bottom, traveled along the bottom strata of water and thus reached the hydrophone rather than by any reflection method.

The results of these comparisons when examined for "primary maxima" simply indicate that the quickest path for the necessary amount of energy to reach the hydrophone was along the low resistance path of the cold water near the bottom, rather than along the upper layers. This hypothesis is still further strengthened by comparisons of measurements made where the depth was 2000 fathoms. The measured velocity in this case was 1479 meters per second while the calculated bottom velocity was 1501 meters per second, the greater velocity being due to the increased pressure. Assuming a path, however, 375 fathoms below the surface, the velocity would be 1477 fathoms which would indicate that at this depth sufficient sound energy would be transmitted to operate the relay.

Assuming that the sound is deflected to the bottom and travels along the lower strata as indicated in the two preceding paragraphs the following point of interest arises. In the survey of Georges Bank the hydrophones are suspended about 8 fathoms below the surface and in some

instances the bottom is as much as 15 fathoms below the hydrophones. The sound from the bomb does actually reach the receiver and the writer asks *how* does the sound know when to bend up again so as to reach the hydrophone?

The determinations of the velocity of sound from explosions by the British in the English Channel[5] show extraordinary care in making their measurements, including their work at neap tide slack water to avoid water currents; the use of several hydrophones approximately in a straight line, and the firing of a series of bombs by a destroyer steaming in a line at right angles to that of the hyrdophones so that the times could be taken between pairs of hydrophones exactly in line with the explosion. Times were measured to thousandths of a second on a multiple string galvanometer record. One set of tests was made in September with an average temperature of 17° and another set in February with temperatures of 6° and 7°C. They were extremely careful to get the temperature measurements at the time of the tests and took an average of the temperatures at the bottom, mid-depth, and six feet below the surface to represent the temperature at that position and they found that "the vertical temperature gradient was generally negligible. . . . Owing to the lack of homogeneity of the surface layers temperature observations were not made at depths less than six feet."

They fired bombs of 18½ pounds of gun cotton at depths of 10, 40 and 108 feet, and 300 pounds at 50 feet. The distances were roughly 4, 8 and 11 miles between hydrophones and a visual average of the depths by inspection of their chart would be 7 to 20 fathoms.

Their conclusions were that no certain differences in velocity were observed between explosions of detonators and other size charges up to 300 pounds. No difference was detectable through the different depths of explosions or the different depths of water between hydrophones (which was slight).

It should be noted that the timing device used in their method did not involve mass motion comparable to that of a relay or chronograph; also that the British results are based on the assumption that sound passes through the entire cross section (excepting 6 feet at the surface) which would be in agreement with the present hypothesis of transmission. That is, for the "primary maximum" the sound energy factor would be negligible and the average velocity would agree with the measured value.

Many text books and papers on acoustics quote the results of Threl-

[5] Proceedings of the Royal Society, Vol. 103 A, 1923.

fall and Adair[6] in which they apparently obtained velocities of 2000 meters per second for large charges of gun cotton and a lower velocity for smaller charges. The writer never believed that a loud sound could travel faster than a weak one and after a critical study of their elaborate experiments, which were made in Australia in 1889, believes that the contacts on their receivers chattered on the receipt of strong signals, thereby sending a pulsating current rather than a steady flow to their magnets at the beginning of the records consequently causing the markers to function improperly. Their distances were only about 500 feet and consequently the time intervals were only about a tenth second so that any errors in time made large percentage errors. Careful reading of parts of their work indicates that possibly they themselves doubted the results, but felt compelled to abide by the apparent evidence. Their receivers were submerged drums having soft rubber heads which sent a compressional wave of air to a second rubber diaphragm actuating the contacts. The published records show about 100 pulsations per second and the faint marks which appear on the smoked glass tracings which they thought might be due to echoes are believed to be simply the beginnings of true records. A chart showing the region of their work was studied to determine what effect echoes might have had on their results. However, it does not seem possible to coordinate any echo time values with those recorded, nor does it seem possible to decide from their published records what values might be used to give normal sound velocities, consequently, the writer believes it was simply an elaborate piece of experimental work from which erroneous conclusions were drawn.

The fathometer, when used in depths of, say 10 fathoms, will indicate shallower depths for a strong signal on the oscillator than for a weak signal, and one might conclude that the velocity was greater for a strong signal than for a weak one, but fortunately such is not the case. The fathometer will indicate exactly similar results if the intensity of the received signal is varied by just changing the input potentiometer. In the equation "force equals mass times acceleration," the acceleration is directly proportional to the force and this appears true in electrical circuits as well as in mass motion so that if a strong signal reaches the amplifier the red light appears quicker than it will for a weak one. This effect is not ordinarily so noticeable on the 412 type of fathometer for the energy of the blow from the striker is much greater than that given by the oscillator. However, with considerable amplification available at 60 fathoms, the same effect was observed. It might be mentioned for

[6] Proc. Royal Soc., London, Vol. 46, 1889.

those who are not familiar with the fathometer that the variation in time here mentioned is only about 0.003 second.

It has been observed on some of the Coast Survey ships on the West Coast and reported in the article,[4] that when depths of 200 fathoms and more are calculated by the aid of bombs and chronograph records, it appears that the velocity of sound is greater for the first echo from the bomb than for succeeding ones and also is greater than for the sound of the oscillator used in echo sounding. It is believed, however, that this apparent discrepancy is due entirely to the quicker response of the time measuring apparatus to the greater intensity of the bomb signal than to the weaker oscillator signal.

During regular surveying on Georges Bank, the writer made two records where a half pint of TNT and a detonator were fired within a few seconds of each other. In the first instance, the detonator gave a deflection of 2 milliamperes and the time elapsed was 20.17 seconds, while the half pint of TNT gave a deflection of 10 milliamps and the time was 20.17 seconds. The two were dropped overboard simultaneously and the detonator exploded just 3 seconds before the half pint of TNT. In the second instance the half pint gave a deflection of 10 milliamperes and the time was 20.22 seconds while the detonator gave a deflection of 1 milliampere and the time was 20.23 seconds. Again the two were dropped together and the half pint fired just 7 seconds before the detonator.

The difference of 0.01 second in the latter record is no greater than the probable accuracy of measuring the tape and these two records should be sufficient proof that the strong and weak signals travel with the same velocity even if there were not the evidence of the British experiments. There is also the evidence from the above tests that the sound energy received from the smaller explosion has sufficient intensity to actuate the recording apparatus just as quickly as does the larger explosion.

When opportunity permits the writer plans making a detailed study of subaqueous sounds by oscilligraphic curves. It is hoped that considerable information will thereby be gained in this interesting subject.

SUMMARY

The main points of the preceding discussion may be summarized as follows:

1. The hypothesis of transmission of sound through sea water by multiple reflection does not seem justifiable. Instead, a hypothesis is

[4] Loc. Cit.

proposed in which the method of tramsmission would not be far different from the transmission of sound through air.

2. Sub-aqueous sounds spread out in all directions unless constrained.

3. The ordinary laws of reflection apply.

4. The velocity of sound is greater in warm water than in cold.

5. The acoustic resistance of sea water is greater when warm than when cold.

6. The time of travel of a sub-aqueous sound from one point to another, is, in general, not a single definite amount, but may be a series of values of relatively considerable length.

7. The measured time of travel will depend upon the measuring apparatus, being less for more sensitive instruments.

8. In R.A.R. work the peak of sound energy which operates the radio signal may not be the first impulse to arrive.

9. The velocity of transmission is independent of the intensity of the signal, but recording instruments may give erroneous values.

BOOK REVIEW

Planning for good acoustics, by HOPE BAGENAL (Lecturer in the Acoustics of Buildings at the Architectural Association School) and ALEX. WOOD (Lecturer in Experimental Physics in the Univ. of Cambridge). Pp. 415. Metheun & Co., London, 1931, Price 21s.

This wook is a general text-book on the acoustics of buildings; not only from the scientific viewpoint, but more particularly from that of the architect. It includes 16 chapters, 4 appendices and 236 illustrations and plans. The first two chapters deal with wave motion and reverberation; chapters 3–8 give discussions of plans and sections of buildings designed for concert halls, parliaments, theaters, residences, hospitals, churches, etc.; chapter 9 is concerned with sound absorbents and their application; chapter 10 deals with sound recording studios; chapter 11, with acoustic design in tropical countries; chapter 12, with sound transmission; chapter 13, with sound-proofing and prevention of noise; chapter 14, with influence of ventilation on acoustics; chapter 15 gives a summary of practical suggestions for architects, and chapter 16 concludes with a historical development of buildings planned acoustically.

As indicated by the title of the book, the attempt is made throughout to coordinate the science of acoustics with the design of buildings. Tables of data of important scientific investifations are reproduced; they are not merely quoted, but are subjected to discussion based on a wide reading and study of published articles. The authors set forth opinions about acoustic theories and phenomena that are in general agreement with those of other acousticians, but they also give original ideas that make the book a contribution to the subject. The main purpose of the book, however, is to apply the published investigations to the problem of planning good acoustics in buildings. In accordance with this purpose, a large number of photographs and plans of various auditoriums are published, with calculations and comments on the acoustic features. An architect interested in securing a suitably designed room with good acoustics will find a wealth of material for his guidance.

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